Exhibit C

U.S. Patent Number 4,910,779 of Cooper (1990)

United States Patent [19]

Cooper et al.

[11] **Patent Number: 4,910,779**

[45] **Date of Patent: Mar. 20, 1990**

[54] **HEAD DIFFRACTION COMPENSATED STEREO SYSTEM WITH OPTIMAL EQUALIZATION**

[76] Inventors: **Duane H. Cooper**, 918 W. Daniel St., Champaign, Ill. 61821; **Jerald L. Bauck**, 1007 W. Clark, #4, Urbana, Ill. 61801

[21] Appl. No.: **266,139**

[22] Filed: **Nov. 2, 1988**

Related U.S. Application Data

[63] Continuation-in-part of Ser. No. 109,197, Oct. 15, 1987.

[51] **Int. Cl.**[4] **H04S 1/00**

[52] **U.S. Cl.** **318/26**; 381/1

[58] **Field of Search** 381/1, 17, 18, 19, 20, 381/21, 22, 23, 24, 25, 26

[56] **References Cited**

U.S. PATENT DOCUMENTS

3,088,997 5/1963 Bauer .
3,236,949 2/1966 Atal et al. .
3,238,304 3/1966 Yaita et al. .
3,566,294 2/1971 Takahashi .
3,710,023 1/1973 Greuzard, Sr. et al. .
3,725,586 4/1973 Iida .
3,892,624 7/1975 Shimada .
4,058,675 11/1977 Kobayashi et al. .
4,118,599 10/1978 Iwahara et al. .
4,119,798 10/1978 Iwahara .
4,136,260 1/1979 Asahi .
4,139,728 2/1979 Haramoto et al. .
4,192,969 3/1980 Iwahara .
4,199,658 4/1980 Iwahara .
4,218,585 8/1980 Carver .
4,219,696 8/1980 Kogure et al. .
4,308,423 12/1981 Cohen .
4,356,349 10/1982 Robinson .
4,359,605 11/1982 Haramoto et al. .
4,388,494 6/1983 Schone et al. .
4,489,432 12/1984 Polk .
4,503,554 3/1985 Davis .
4,566,119 1/1986 Peters .
4,731,848 3/1988 Kendall et al. .
4,748,669 5/1988 Kalyman .
4,819,269 4/1989 Klayman .

FOREIGN PATENT DOCUMENTS

1269187 5/1968 Australia .
2406712 8/1975 Fed. Rep. of Germany .
2308267 4/1976 France .
394325 7/1933 United Kingdom .
781186 8/1957 United Kingdom .
1367705 8/1971 United Kingdom .
1425519 2/1976 United Kingdom .
1459188 12/1976 United Kingdom .

OTHER PUBLICATIONS

Cooper, Duane H., "Calculator Program for Head-Related Transfer Function", Audio Engineering Society, Vol. 30, No. ½, 1982 Jan./Feb., pp. 34–38.
"Controlling Sound-Image Localization in Stereophonic Reproduction", J. Audio Engineering Society, Vol. 29, No. 11, 1981, Nov., pp. 794–799.
"Controlling Sound-Image Localization in Stereo-

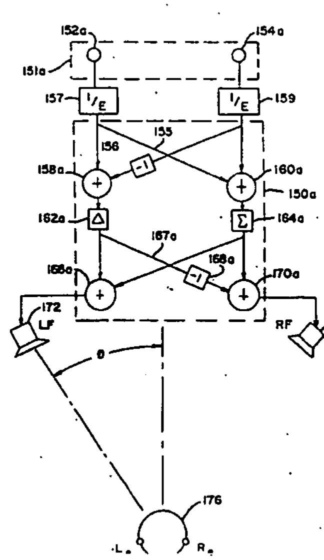
(List continued on next page.)

Primary Examiner—Forester W. Isen
Attorney, Agent, or Firm—Welsh & Katz, Ltd.

[57] **ABSTRACT**

A stereo audio processing system for a stereo audio signal processing reproduction that provides improved source imaging and simulation of desired listening environment accoustics while retaining relative independence of listener movement. The system first utilizes a synthetic or artificial head microphone pickup and utilizes the results as inputs to an equalization circuit with the outputs coupled to a cross-talk cancellation compensation circuit utilizing minimum phase filter circuits to adapt the head diffraction compensated signals for use as loudspeaker signals. The system provides for head diffraction compensation including equalization and cross-coupling while permitting listener movement by modifying the cross-talk cancellation and diffraction compensation at frequencies substantially above approximately ten kilohertz while maintaining substantially accurate equalization for the desired incidence angle.

14 Claims, 16 Drawing Sheets



OTHER PUBLICATIONS

phonic Reproduction: Part II*", J. Audio Eng. Soc., Vol. 30, No. 10, 1982, Oct., pp. 719–723.
"Precision Sound-Image-Localization Technique Utilizing Multitrack Tape Masters*", Audio Eng. Soc., Vol. 27, No. ½, 1979 Jan./Feb., pp. 32–39.
"On the Simulation of Sound Localization", J. Accoust. Soc., Jpn (E) 1, 3 (1980), pp. 167–174.
Bartlett, Bruce, "Recording Techniques: Simple Stereo Microphone Techniques", db Sep.–Oct., 1986.
"On Acoustical Specification of Natural Stereo Imaging", An Audio Engineering Society Reprint (Presented at the 66th Convention, 1980, May 6–9, Los Angeles), 1649 (H–7), pp. 1–53.
Schwarz, Von. L., "Zur Theorie der Beugung einer eben Schallwelle an der Kugel", *Akustische Zeitschrift*, 1943, pp. 91–117.
Schroeder, M. R. et al., "Computer Simulation of Sound Transmission in Rooms", IEEE Conv. Record, pt. 7, pp. 150–155 (1963).
Schroeder, M. R., "Digital Simulation of Sound Transmission in Reverberant Spaces", J. Acoust. Soc. Am., Vol. 47, pp. 424–431 (Feb. 1970).
Schroeder, M. R., "Computer Models for Concert Hall Acoustics", Am. Journal Phys., Vol. 41, pp. 461–471 (April, 1973).
Schroeder, M. R. , "Models of Hearing", Proc. IEEE, Vol. 63, pp. 1332–1350 (Sep., 1975).
Damaske, P., "Head-Related Two-Channel Stereophony with Loudspeaker Reproduction", J. Acoust. Soc. Am., Vol. 50, pt. 2, pp. 1109–1115 (Oct., 1971).
Mehrgardt, S., et al., "Transformation Characteristics of the External Human Ear", J. Acoust. Soc., Am., Vol. 61, pp. 1567–1476 (Jun., 1977).
Cooper, H. et al., "Corrections to L. Schwarz, 'On the Theory of Diffraction of a Plane Soundwave Around a Sphere', ('Zur Theorie der Beugung einer ebenen Schallwelle an der Kugel', Akust Z 8, 91–119 (1943))", J. Acoust. Soc. Am., Vol. 80, pp 1793–1802 (Dec., 1986).
Gerzon, M. A., "Stereo Shuffling: New Approach—Old Technique", Studio Sound, pp. 123–130 (Jul., 1986).
Parsons, T. W., "Super Stereo: Wave of the Future?", The Audio Amateur, Vol. IX, pp. 19–20 (Jun., 1978).
Nakabayashi, K., "A Method of Analyzing the Quadraphonic Sound Field", J. Audio Eng. Soc., Vol. 23, pp. 187–193 (Apr., 1975).

FIG. 1C

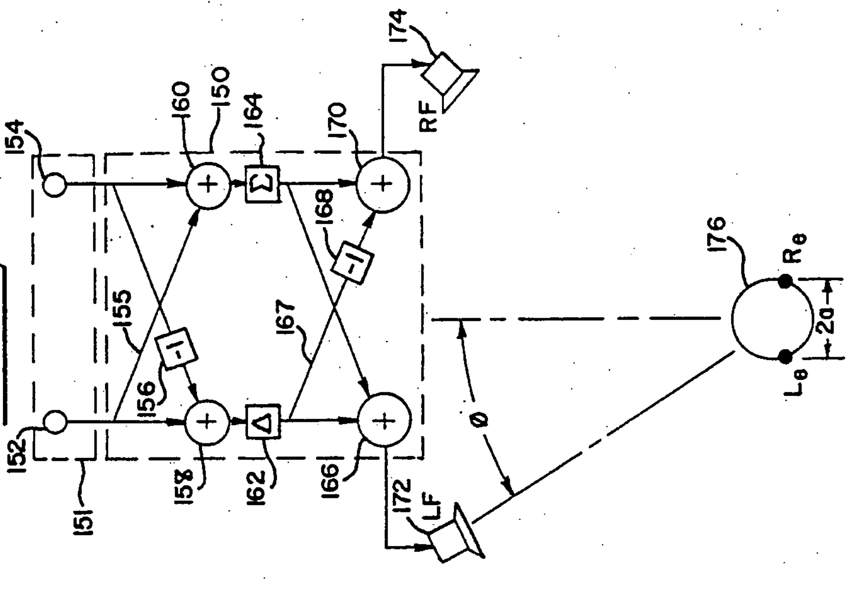

150
151
152
154
155
156
158
160
162
164
166
167
168
170
172
174
176
Σ
Δ
-1
-1
LF
RF
Ø
2a
L_e
R_e

FIG. 1A

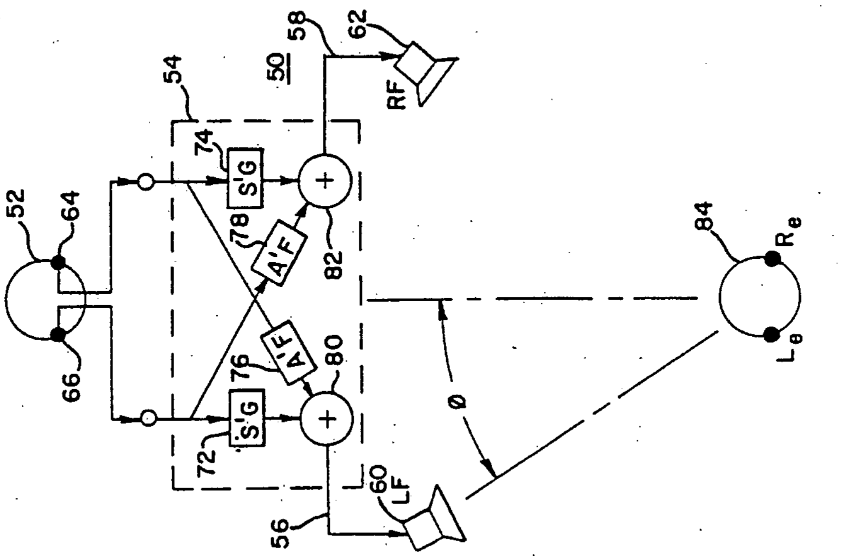

50
52
54
56
58
60
62
64
66
72
74
76
78
80
82
84
S'G
S'G
A'F
A'F
LF
RF
Ø
L_e
R_e

FIG. 1B

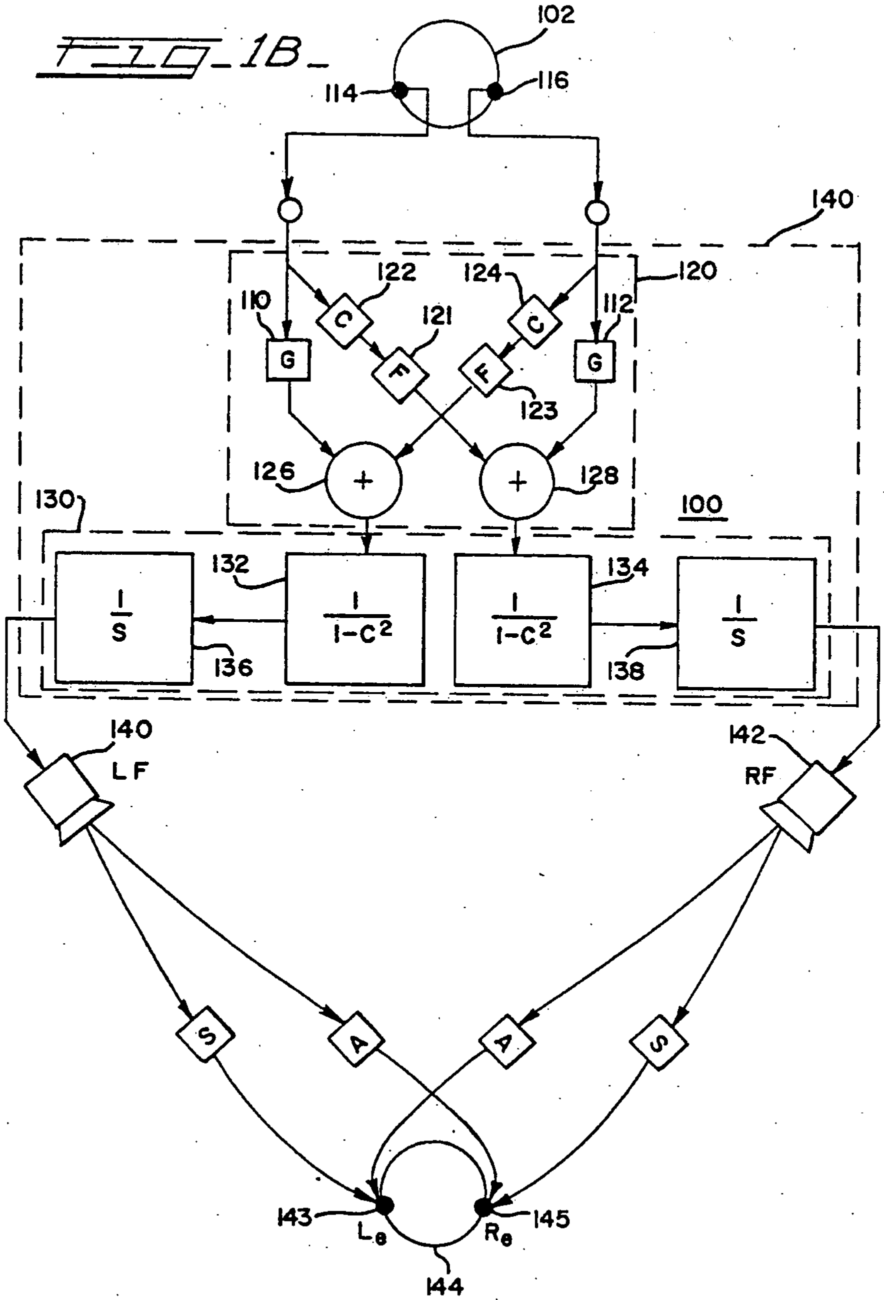

102
114
116
140
120
122
124
110
112
121
C
C
G
G
F
F
123
126
128
+
+
130
100
132
134
1/S
1/(1-C²)
1/(1-C²)
1/S
136
138
140
LF
142
RF
S
A
A
S
143
145
Le
Re
144

FIG. 1D

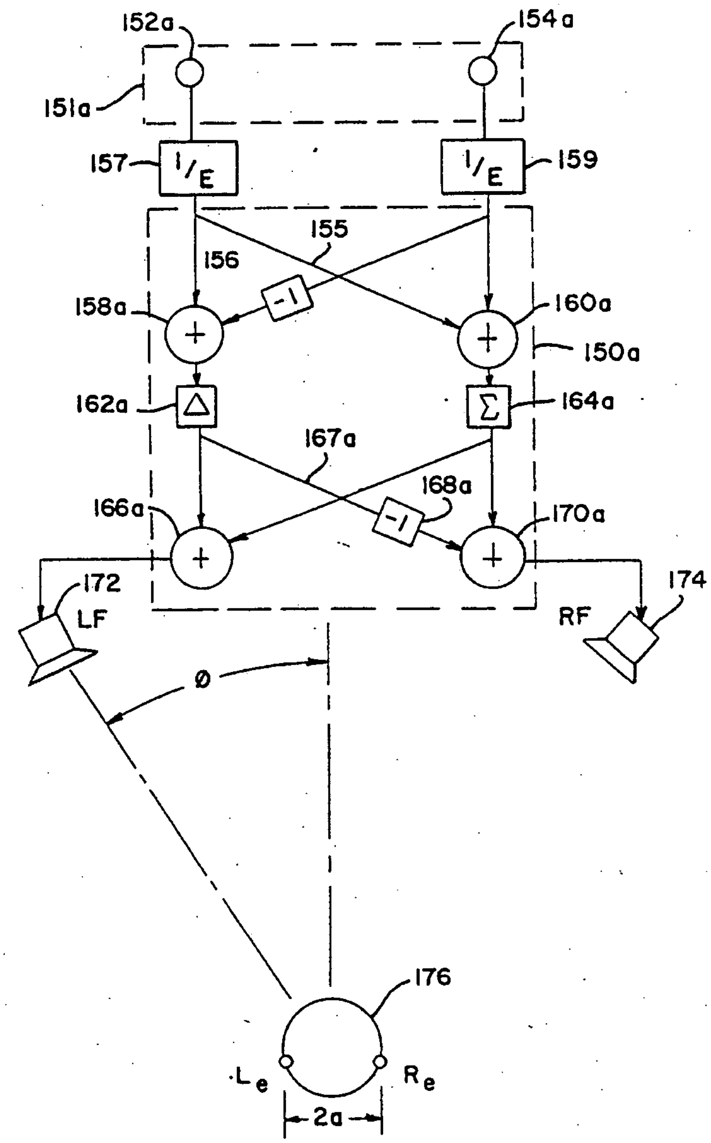

152a
154a
151a
157
1/E
1/E
159
155
156
-1
158a
160a
150a
162a
164a
167a
168a
166a
-1
170a
172
LF
174
RF
Ø
176
L_e
R_e
2a

FIG-2B

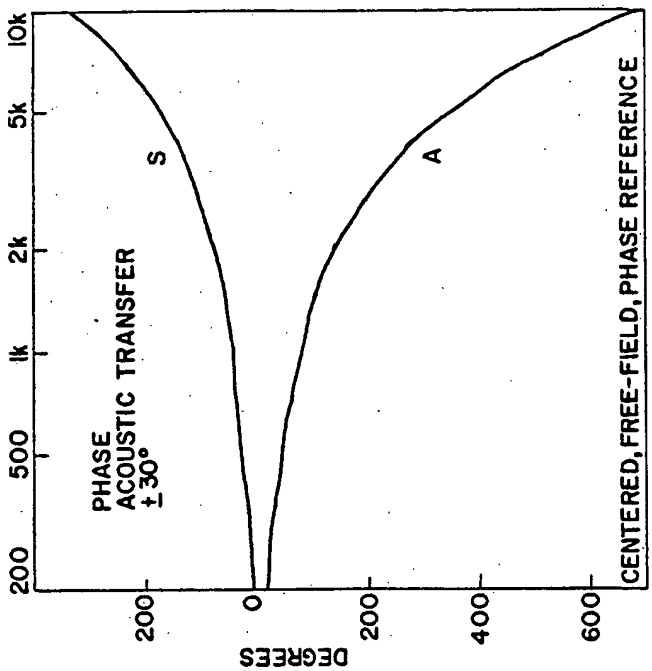

PHASE
ACOUSTIC TRANSFER
±30°
S
A
CENTERED, FREE-FIELD, PHASE REFERENCE
DEGREES
200
0
200
400
600
200
500
1k
2k
5k
10k

FIG-2A

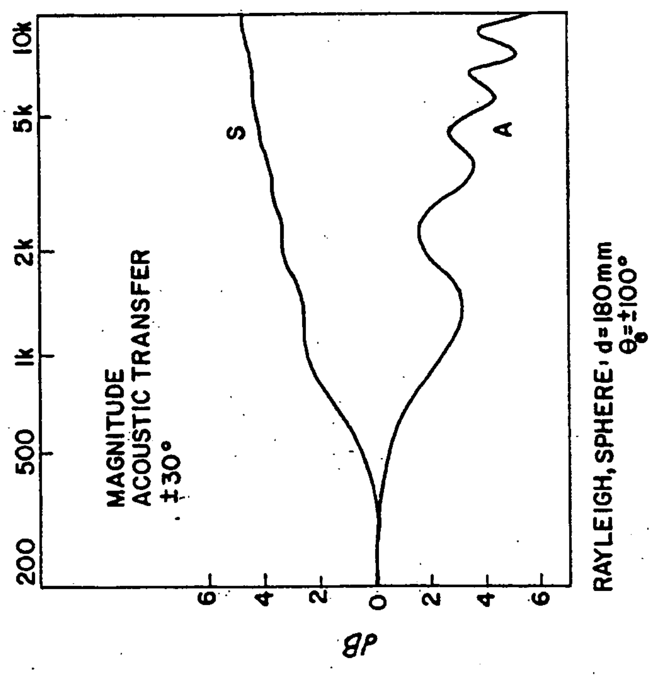

MAGNITUDE
ACOUSTIC TRANSFER
±30°
S
A
RAYLEIGH, SPHERE: d=180mm
θ_e=±100°
dB
6
4
2
0
2
4
6
200
500
1k
2k
5k
10k

FIG. 2D

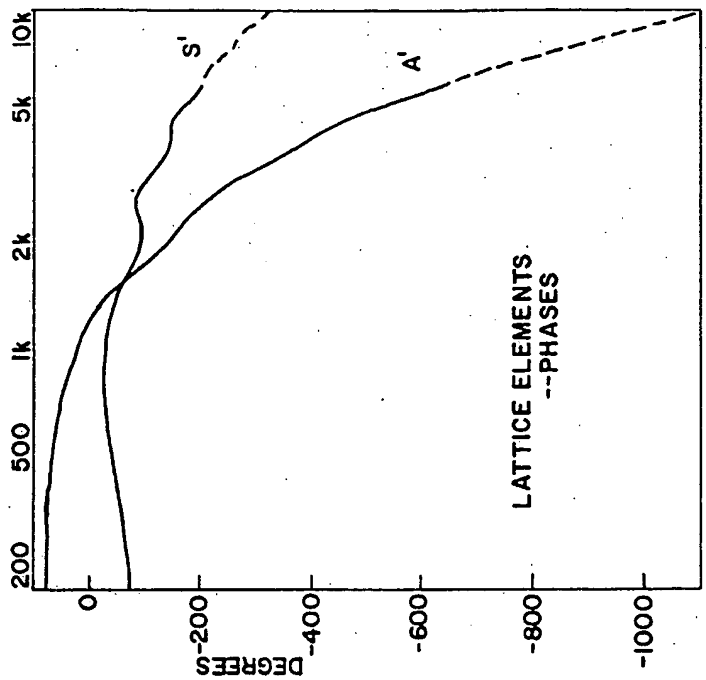

LATTICE ELEMENTS
--PHASES
DEGREES
0
-200
-400
-600
-800
-1000
200
500
1k
2k
5k
10k
S'
A'

FIG. 2C

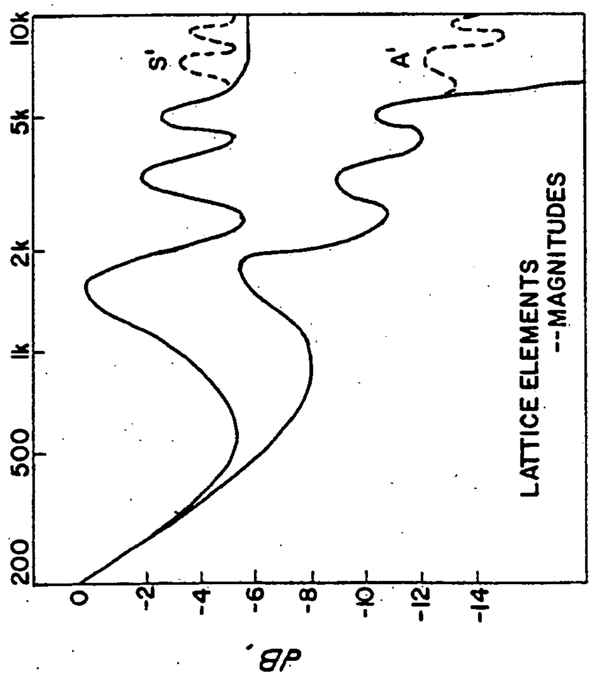

LATTICE ELEMENTS
--MAGNITUDES
dB.
0
-2
-4
-6
-8
-10
-12
-14
200
500
1k
2k
5k
10k
S'
A'

FIG. 3B

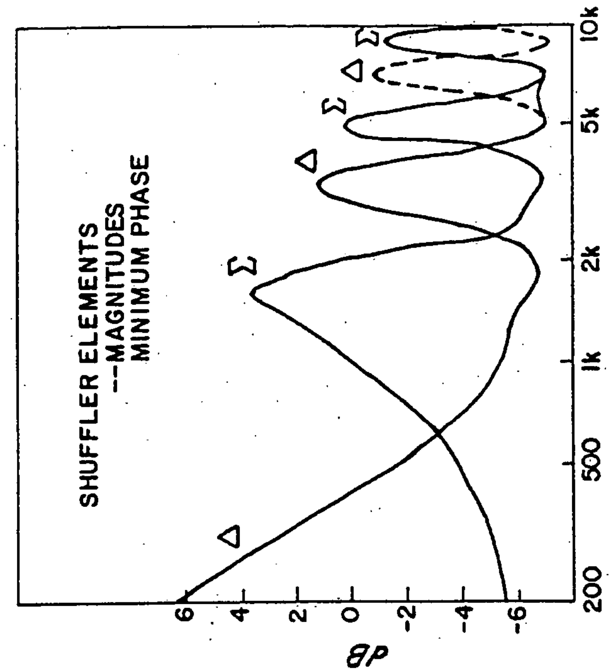

SHUFFLER ELEMENTS
--MAGNITUDES
MINIMUM PHASE
Δ
Σ
Δ
Σ
Δ
Σ
dB
6
4
2
0
-2
-4
-6
200
500
1k
2k
5k
10k

FIG. 3A

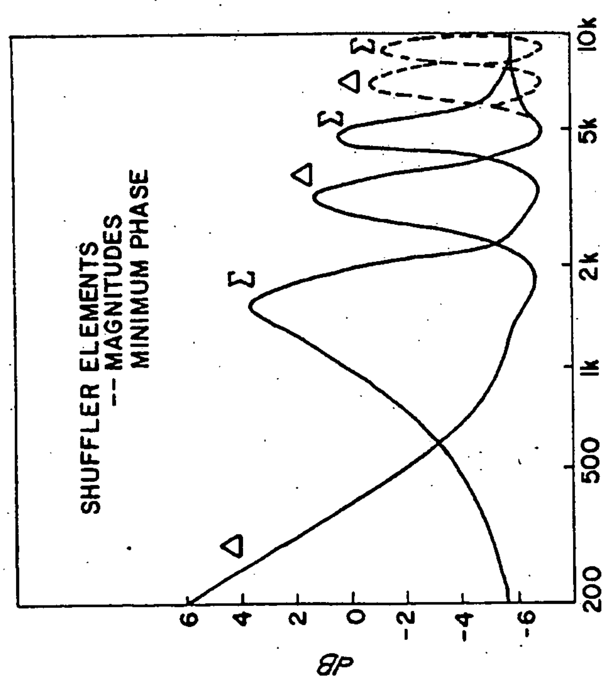

SHUFFLER ELEMENTS
--MAGNITUDES
MINIMUM PHASE
Δ
Σ
Δ
Σ
Δ
Σ
dB
6
4
2
0
-2
-4
-6
200
500
1k
2k
5k
10k

FIG-4A

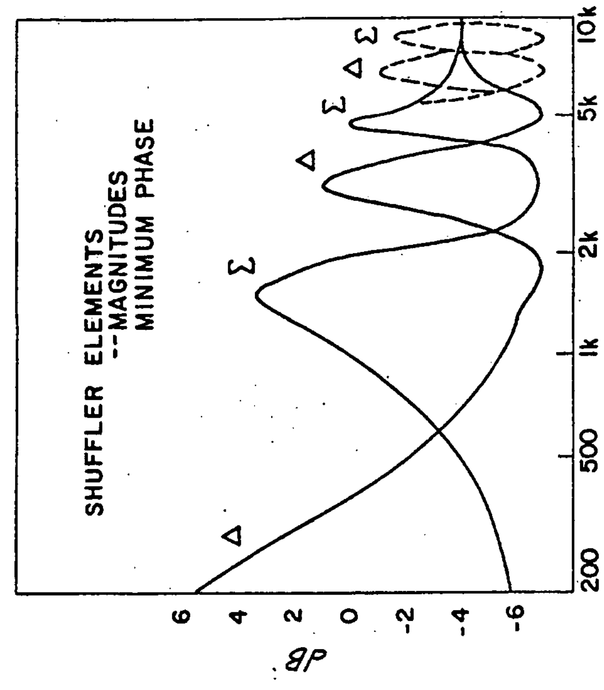

SHUFFLER ELEMENTS
--MAGNITUDES
MINIMUM PHASE
Δ
Σ
Δ
Σ
Δ
Σ
6
4
2
0
-2
-4
-6
dB
200
500
1k
2k
5k
10k

FIG-3C

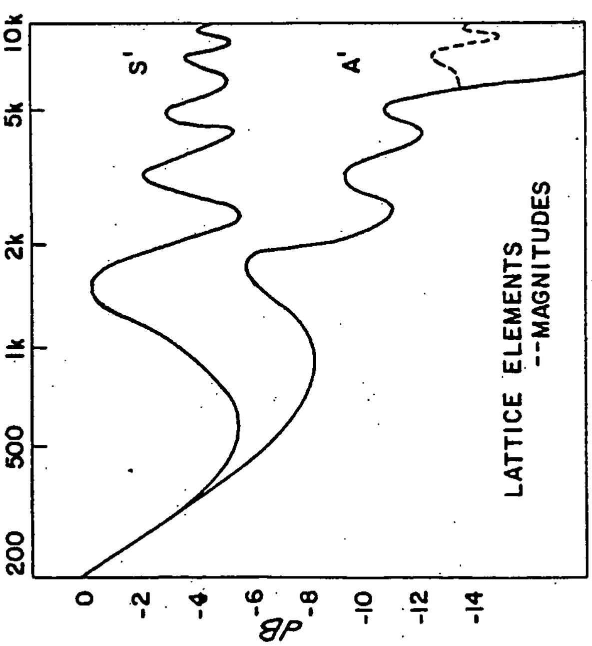

S'
A'
LATTICE ELEMENTS
--MAGNITUDES
0
-2
-4
-6
-8
-10
-12
-14
dB
200
500
1k
2k
5k
10k

FIG-4C

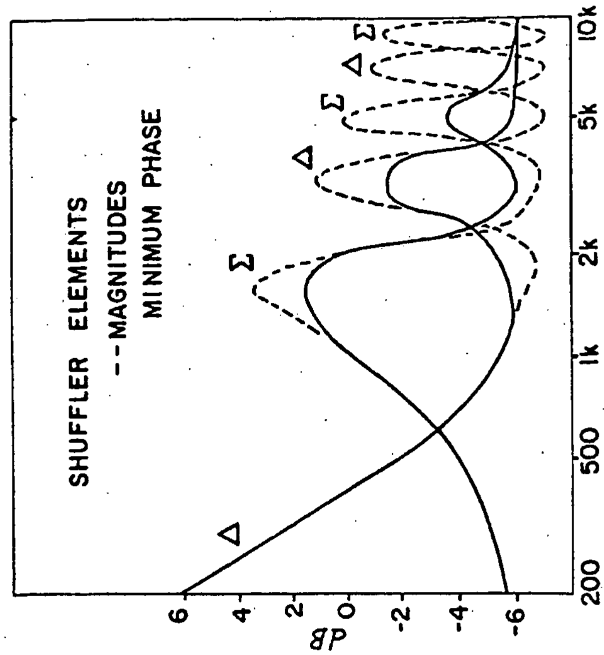

SHUFFLER ELEMENTS
--MAGNITUDES
MINIMUM PHASE
Δ
Σ
Δ
Σ
Δ
Σ
dB
6
4
2
0
-2
-4
-6
200
500
1k
2k
5k
10k

FIG-4B

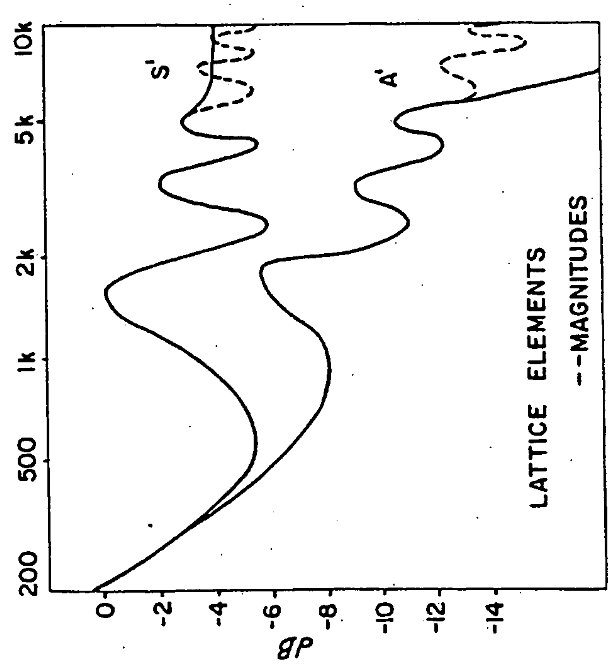

LATTICE ELEMENTS
--MAGNITUDES
S'
A'
dB
0
-2
-4
-6
-8
-10
-12
-14
200
500
1k
2k
5k
10k

FIG-5A

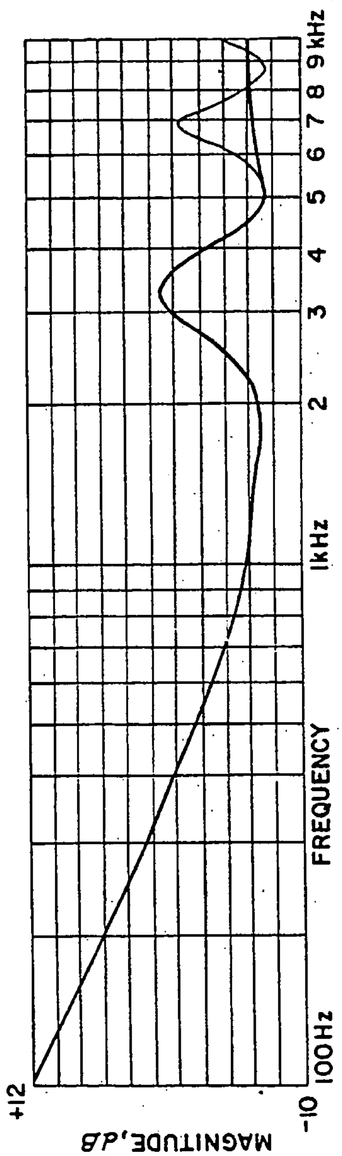
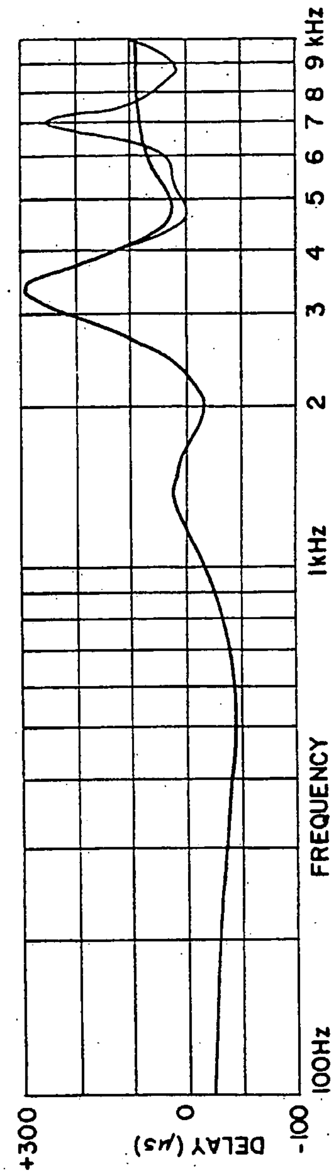

9 kHz
8
7
6
5
4
3
2
1kHz
FREQUENCY
100Hz
+12
-10
MAGNITUDE, dB

FIG-5B


9 kHz
8
7
6
5
4
3
2
1kHz
FREQUENCY
100Hz
+300
0
-100
DELAY (μs)

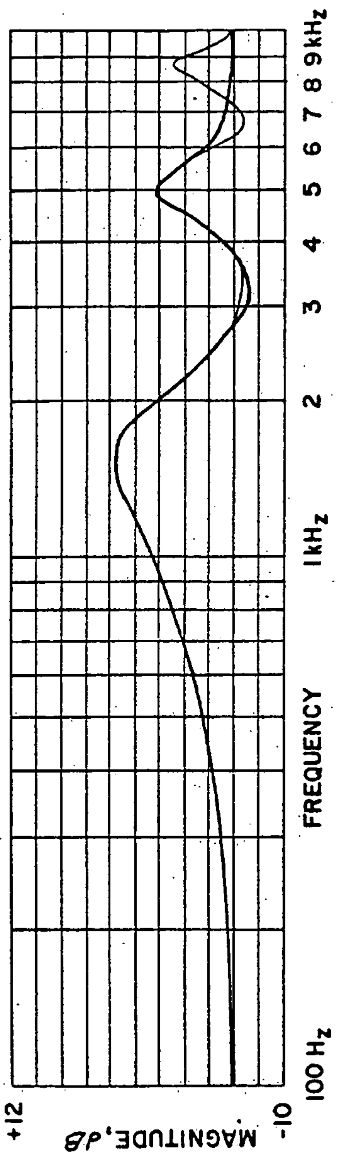

+12
MAGNITUDE, dB
-10
100 Hz
FREQUENCY
1kHz
2
3
4
5
6
7
8
9kHz

FIG-5C

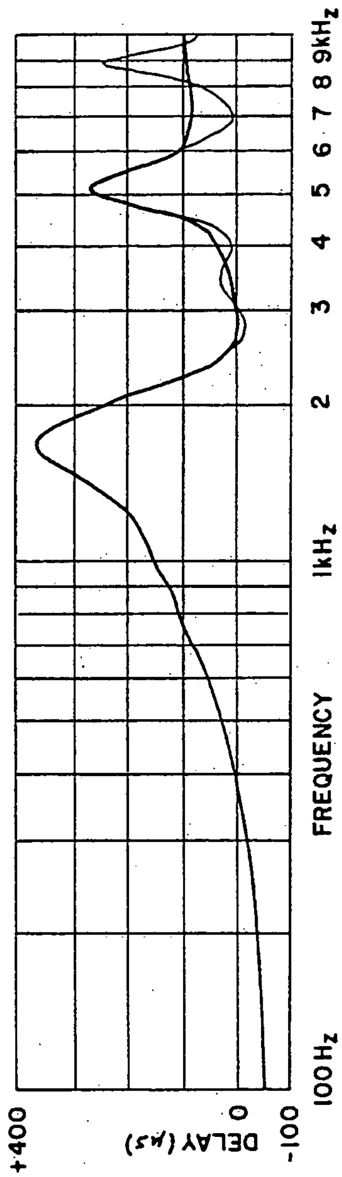

+400
DELAY (μs)
0
-100
100 Hz
FREQUENCY
1kHz
2
3
4
5
6
7
8
9kHz

FIG-5D

FIG. 6

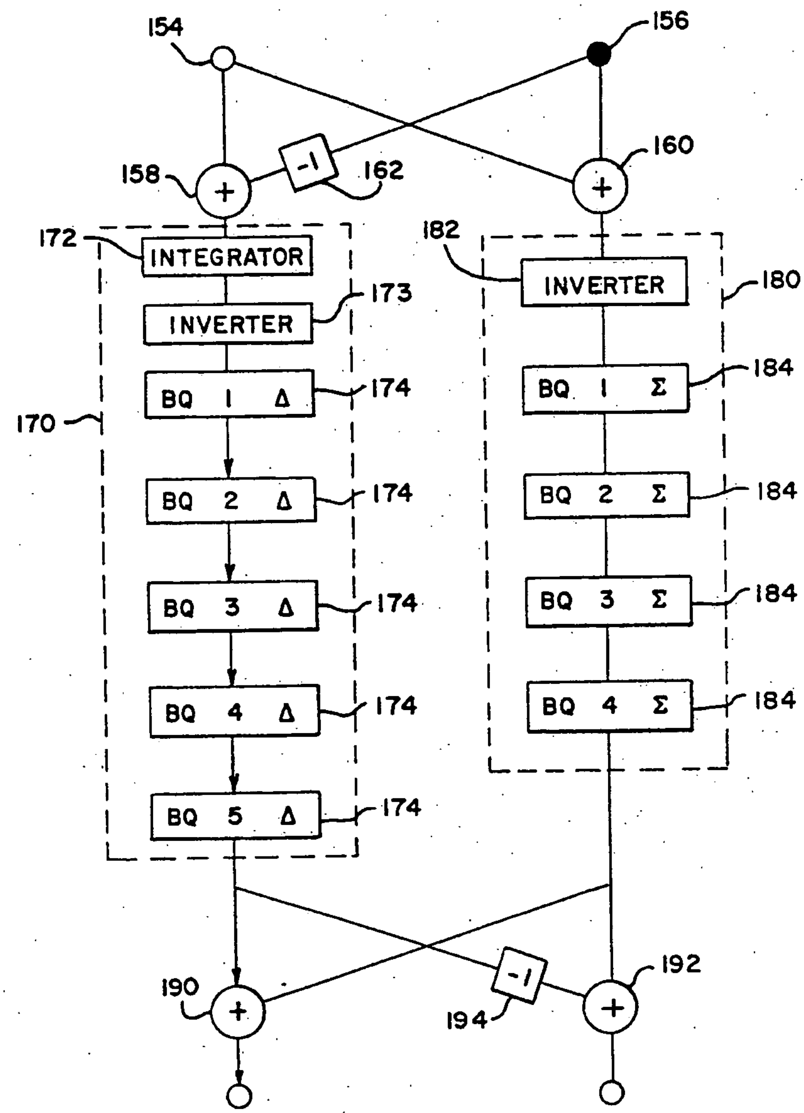

154
156
158
160
-1
162
172
INTEGRATOR
182
INVERTER
180
INVERTER
173
BQ 1 Δ
174
170
BQ 1 Σ
184
BQ 2 Δ
BQ 2 Σ
BQ 3 Δ
BQ 3 Σ
BQ 4 Δ
BQ 4 Σ
BQ 5 Δ
190
192
-1
194

FIG_7_

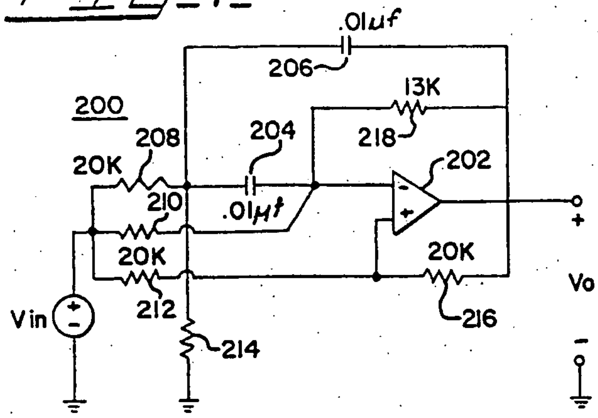


FIG_8A_

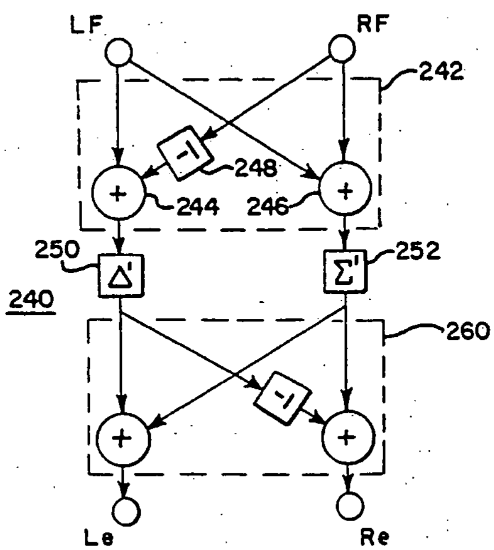


FIG_8B_

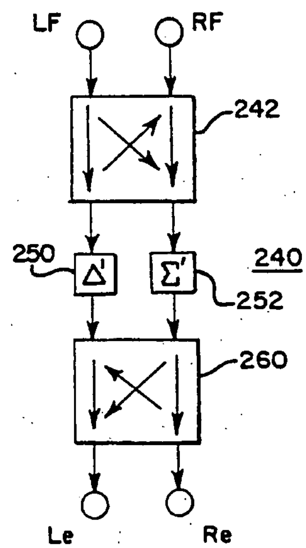
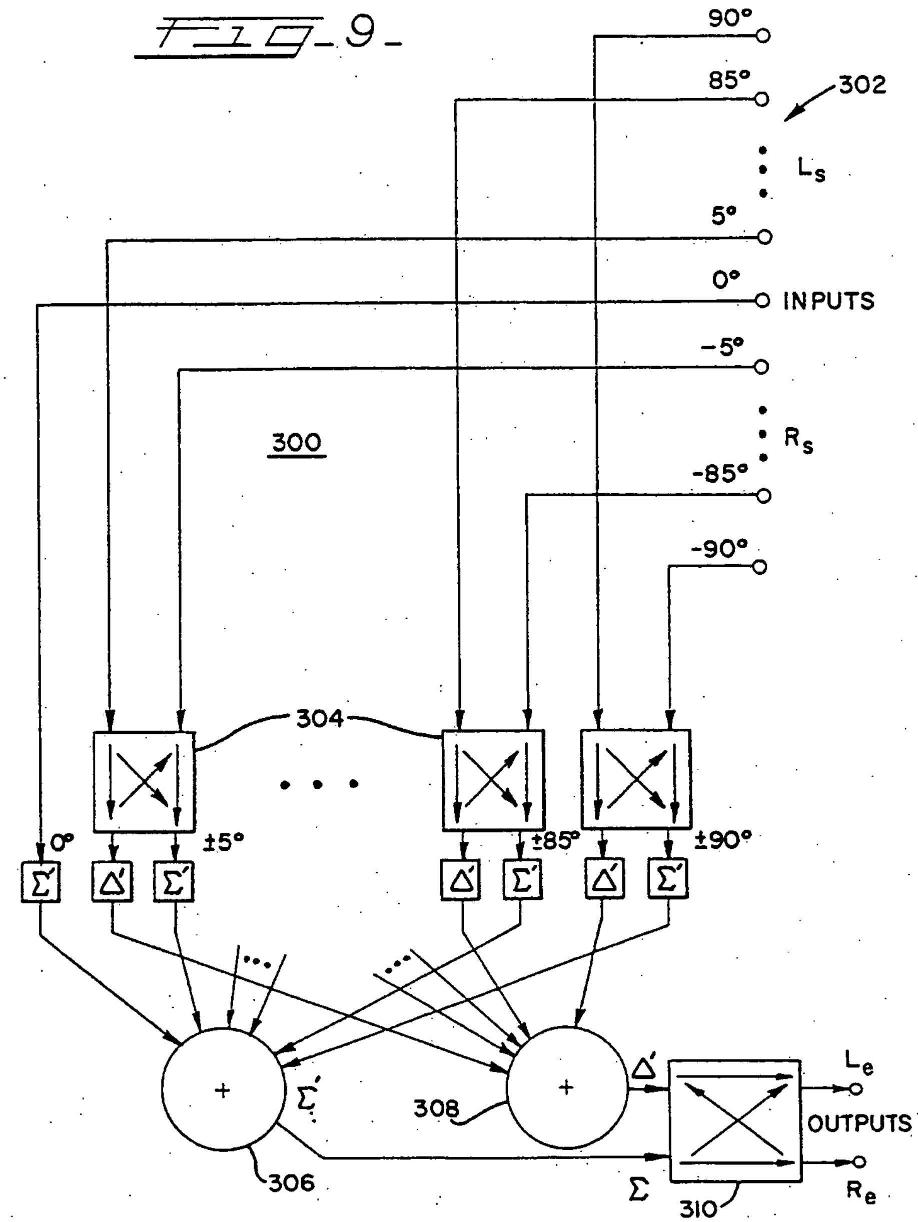


FIG. 9



90°
85°
302
L_s
5°
0°
INPUTS
-5°
300
R_s
-85°
-90°
304
0°
±5°
±85°
±90°
Σ'
Δ'
Σ'
Δ'
Σ'
Δ'
Σ'
+
+
Σ'
Δ'
308
306
Σ
310
L_e
OUTPUTS
R_e

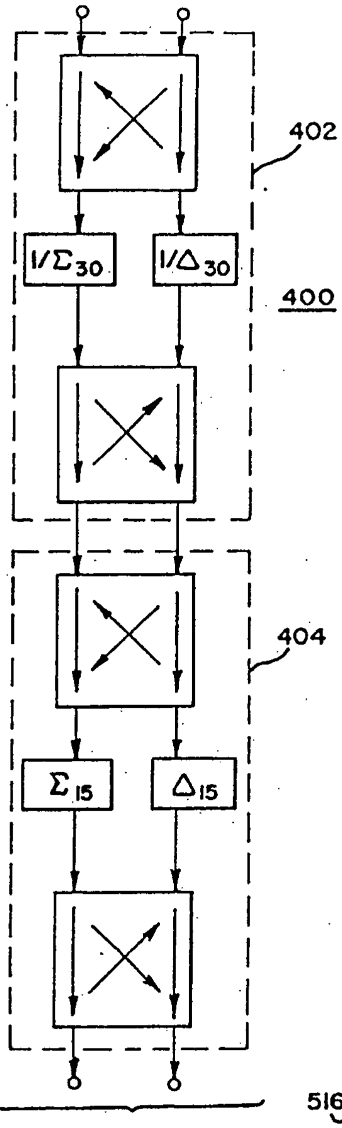


Fig. 10a

Fig. 10b

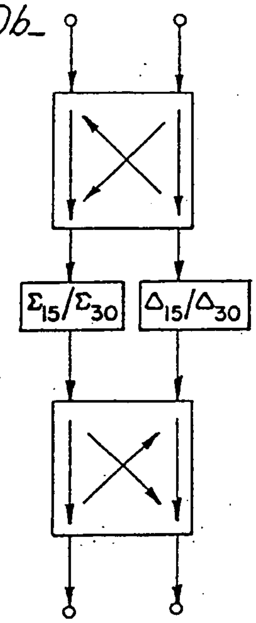


Fig. 11

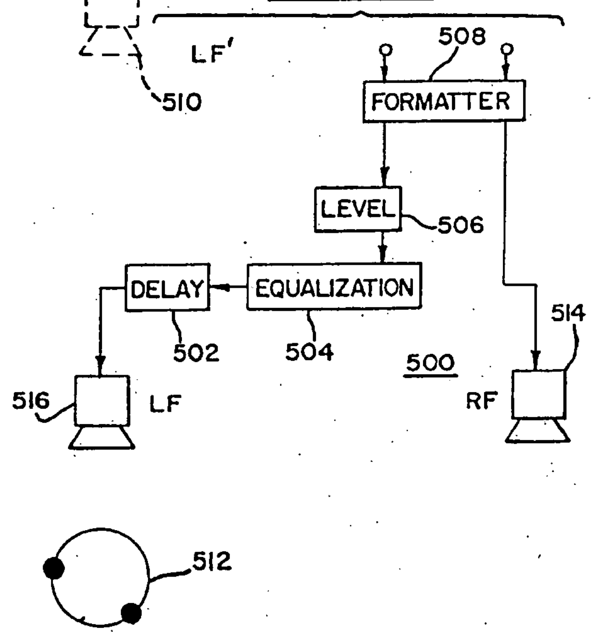


FIG. 12A

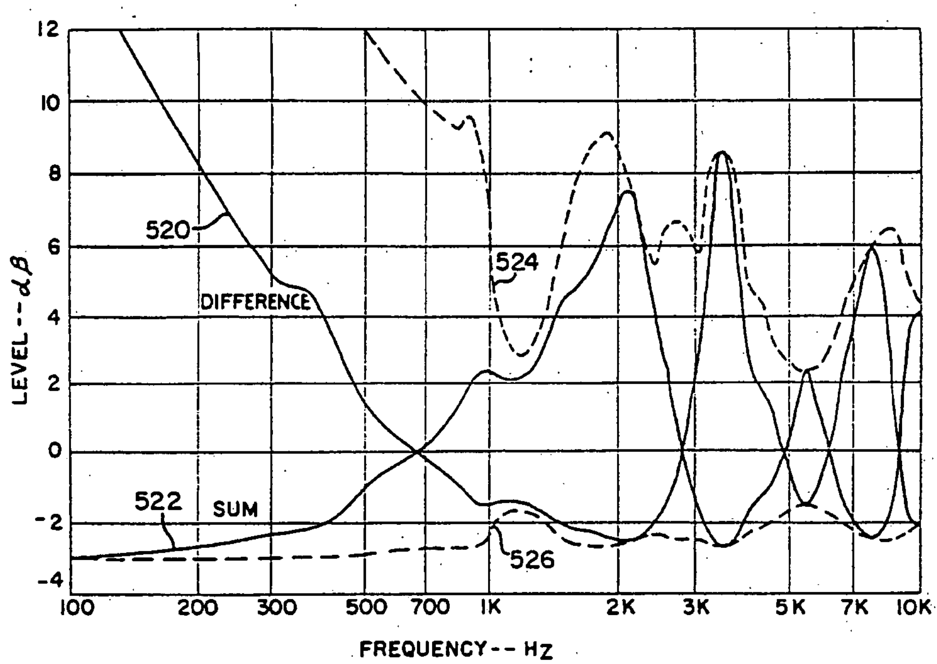
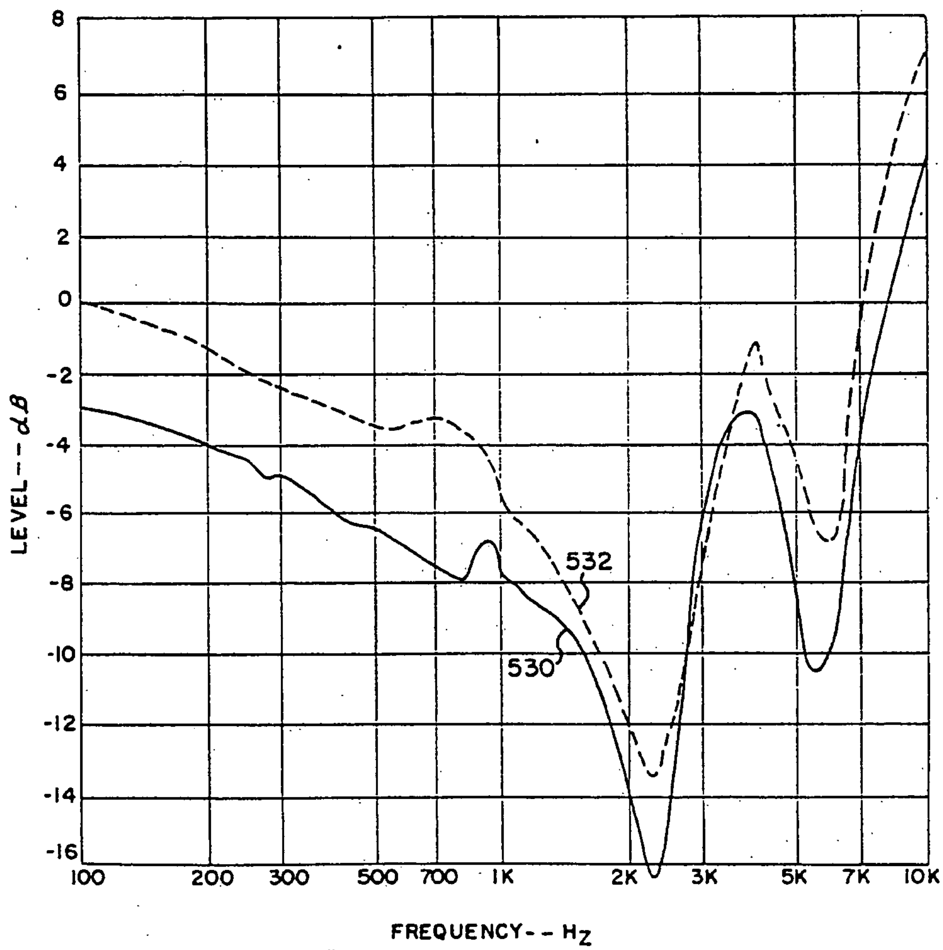

12
10
8
6
4
2
0
-2
-4
LEVEL -- dB
520
DIFFERENCE
524
522
SUM
526
100
200
300
500
700
1K
2K
3K
5K
7K
10K
FREQUENCY -- HZ

FIG. 12B


8
6
4
2
0
-2
-4
-6
-8
-10
-12
-14
-16
LEVEL--dB
532
530
100
200
300
500
700
1K
2K
3K
5K
7K
10K
FREQUENCY--Hz

HEAD DIFFRACTION COMPENSATED STEREO SYSTEM WITH OPTIMAL EQUALIZATION

BACKGROUND OF THE INVENTION

This is a continuation in part of application Ser. No. 109,197, filed Oct. 15, 1987.

This invention relates generally to the field of audio-signal processing and more particularly to a system and method for stereo audio-signal processing and stereo sound reproduction incorporating head-diffraction compensation, which provides improved sound-source imaging and accurate perception of desired source-environment acoustics and equalization to ensure a natural sound quality under variety of listener-environment conditions while maintaining relative insensitivity to listener position and movement.

There is a wide variety of prior-art stereo systems, most of which fall within three general categories or types of systems. The first type of stereo system utilizes two omnidirectional microphones usually spaced approximately one half to two meters apart and two loudspeakers placed in front of the listener towards his left and right sides in correspondence one for one with the microphones. The signal from each microphone is amplified and transmitted, often via a recording,, through another amplifier to excite its corresponding loudspeaker. The one-for-one correspondence is such that sound sources toward the left side of the pair of microphones are heard predominantly in the left loudspeaker and right sounds in the right. For a multiplicity of sources spread before the microphones, the listener has the impression of a multiplicity of sounds spread before him in the space between the two speakers, although the placement of each source is only approximately conveyed, the images tending to be vague and to cluster around loudspeaker locations.

The second general type of stereo system utilizes two unidirectional microphones spaced as closely as possible, and turned at some angle towards the left for the leftward one and towards the right for the rightward one. The reproduction of the signals is accomplished using a left and right loudspeaker placed in front of the listener with a one-for-one correspondence with the microphones. There is very little difference in timing for the emission of sounds from the loudspeakers compared to the first type of stereo system, but a much more significant difference in loudness because of the directional properties of the angled microphones. Moreover, such difference in loudness translates to a difference in time of arrival, at least for long wavelengths, at the ears of the listener. This is the primary cue at low frequencies upon which human hearing relies for sensing the direction of source. At higher frequencies (i.e., above 600 Hz), directional hearing relies more upon loudness differences at the ears, so that high frequency sounds in such stereo systems have thus given the impression of tending to be more localized close to the loudspeaker positions rather than spread as the original sources had been.

The third general type of stereo system synthesizes an array of stereo sources, by means of electrical dividing networks, whereby each source is represented by a single electrical signal that is additively mixed in predetermined proportions into each of the two stereo loudspeaker channels. The proportion is determined by the angular position to be allocated for each source. The loudspeaker signals have essentially the same characteristic as those of the second type of stereo system.

Based upon these three general types of stereo systems, there are many variants. For example, the first type of system may use more than two microphones and some of these may be unidirectional or even bidirectional, and a mixing means as used in the third type of system may be used to allocate them in various proportions between the loudspeaker channels. Similarly, a system may be primarily of the second type of stereo system and may use a few further microphones placed closed to certain sources for purposes of emphasis with signals to be proportioned between the channels. Another variant of the second type of stereo system makes use of a moderate spacing, for example 150 mm, between the microphones with the left angled microphone spaced to the left, and the right-angle microphone spaced to the right. Another variant uses one omnidirectional microphone coincident, as nearly as possible, with a bidirectional microphone. This is the basic form of the MS (middle-side) microphone technique, in which the sum and difference of the two signals are substantially the same as the individual signals from the usual dual-angled microphones of the second type of system.

Each of these systems has its advantages and disadvantages and tends to be favored and disfavored according to the desires of the user and according to the circumstances of use. Each fails to provide localization cues at frequencies above approximately 600 Hz. Many of the variants represent efforts to counter the disadvantages of a particular system, e.g., to improve the impression of uniform spread, to more clearly emulate the sound imaging, to improve the impression of "space" and "air," etc. Nevertheless, none of these systems adequately reckons with the effects upon a soundwave of propagation in the space close to the head in order to reach the ear canal. This head diffraction substantially alters both the magnitude and phase of the soundwave, and causes each of these characteristics to be altered in a frequency-dependent manner.

The use of head-diffraction compensation to make greatly improved stereo sound in a loudspeaker system was demonstrated by M. R. Schroeder and B. S. Atal to emulate the sounds of various concert halls with extraordinary accuracy. Schroeder measured the values of head-related transfer functions for an artificial or "dummy" head (i.e., a physical replica of a head mounted on a fully-clothed manikin) that had microphones placed in its ear canals. This information was used to process two-channel sound recorded using a second artificial head (i.e., to process a binaural recording). Since each ear hears both speakers, the system used crosstalk cancellation to cancel the effects of sound traveling around the listener's head to the opposite ear. Crosstalk cancellation was performed over the entire audio spectrum (i.e., 20 Hz to 20 KHz)

For a listener whose head reasonably well matched the characteristics of the manikin head, the result was a great improvement in characteristics such as spread, sound-image localization and space impression. However, the listener had to be positioned in an exact "sweet spot" and if the listener turned his head more than approximately ten degrees, or moved more than approximately 6 inches the illusion was destroyed. Thus, the system was far too sensitive to listener position and movement to be utilized as a practical stereo system.

In addition, in the prior art, several equalization doctrines may be found. In one of these, a coupler for fitting microphones into an artificial head provides an acoustic equalization corresponding to a flat ear-drum pressure response. Another doctrine specifies a flat response with respect to a diffuse sound field. These two approaches are indicated in a paper by M. Killion, "Equalization Filter for Eardrum Pressure Recording Using KEMAR Manikin," J. Audio Engr. Soc., vol. 27, pp. 13–16 (1979 Jan./Feb.). Yet another doctrine demands a flat pressure response at the ear-canal entrance, as used in certain known artificial heads (e.g., in the Neumann KU-80). On the other hand, Schone, et al., U.S. Pat. No. 4,338,494, teaches that the microphone response should be equalized flat with reference to a free-field, plane wave, incident at 0°.

The role of the equalization is to remove those frequency characteristics of the artificial head that would be essentially repeated, but should not be, in the listener's head. These are the resonances of the cavities in the external ear, the pinna, and, if included in the artificial hear,, the ear canal. The prior art is not correct, however, for incidence angles greater than 0°. For example, it might be desirable, under some circumstances, to place the loudspeakers so that they provide incidence angles of ±90° at an elevation angle at 45°. The frontal, 0° incidence for free-field equalization in the prior art would then prove to be incorrect.

It is accordingly an object of the invention to provide a novel stereo system which provides enhanced sound-imaging localization which is relatively independent of listener position and movement utilizing a novel equalization.

It is another object of the invention to provide a novel stereo system for adapting sound signals utilizing head-diffraction functions, and crosscoupling with filtering to substantially limit the frequency range of such processing to substantially below approximately ten kilohertz to provide enhanced source imaging and accurate perception of simulated acoustics in such frequency range wherein equalization separate from the crosscoupling is provided.

It is a further object of the invention to provide means of utilizing head-diffraction functions and head-diffraction function related equalization so that they may be simulated by means of simple electrical analog or digital filters, in most cases of the minimum-phase type.

It is a further object of the invention to provide a specific combination of free field signals to be used for respective specific incidence angles and to specify these angles in relation to the angles to be used for loudspeaker placement which combination is to be equalized to make for a flat microphone-signal response specifically for that combination.

It is a further object of the invention to provide an equalization method for modifying the signals to or from a crosstalk compensation means by filtering with an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of the squares of the magnitudes of the acoustic transfer functions utilized for the crosstalk filters.

Briefly, according to one embodiment of the invention, an equalization method is provided for an audio processing system that generates compensated audio signals suitable for reproduction to a listener through a loudspeaker system. The audio processing system includes source means for providing two channels of audio signals having head-related transfer functions imposed thereon, and compensation means for providing an inverse crosstalk characteristic of loudspeaker-to-ear listener transmission paths by employing a two port input, and two port output, cross-coupled filter system having transfer functions which approximately simulate acoustic transfer functions of the propagation paths from a loudspeaker to a first ear of the listener and from the loudspeaker to the second ear of the listener. The equalization method is characterized by the step of modifying signals at both ports of either the input or the output of said compensation means by transmission of each signal through a filter that is essentially the same for each of the signals. The filter simulates an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of squares of the magnitudes of the acoustic transfer functions.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention, together with further objects and advantages thereof, may be understood by reference to the following description taken in conjunction with the accompanying drawings.

FIG. 1A is a generalized block diagram illustrating a specific embodiment of a stereo audio processing system.

FIG. 1B is a generalized block diagram illustrating another specific embodiment of a stereo audio processing system.

FIG. 1C is a generalized block diagram illustrating another specific embodiment of a stereo audio processing system.

FIG. 1D is a generalized block diagram illustrating another specific embodiment of a stereo audio processing system including separate equalization according to the invention.

FIG. 2A is a set of magnitude (dB)-versus-frequency-(log scale) response curves of the transfer characteristics from a loudspeaker at 30° to an ear on the same side, curve, S, and to the alternate ear, curve A, used in explaining the invention.

FIG. 2B is a set of phase-(degrees)-versus-frequency-(log scale) response curves of the transfer characteristics from a loudspeaker at 30° to an ear on the same side, curve S, and to the alternate ear, curve A, used in explaining the invention.

FIG. 2C is a set of magnitude-(dB)-versus frequency-(log scale) response curves of the transfer characteristics of the filters shown in FIG. 1A, filters S' and A', continuing in dashed line, and as modified by the factors G and F, respectively, continuing in solid line, used in explaining the invention.

FIG. 2D is a set of phase-(degrees)-versus-frequency-(log scale) response curves of the transfer characteristics of the filters shown in FIG. 1A, filters S' and A', but omitting the phase consequences of the factors G and F, and showing in dashed line the frequency region in which the magnitude modifications are made, used in explaining the invention.

FIG. 3A is a set of magnitude-(dB)-versus frequency-(log scale) response curves of the transfer characteristics of a specific embodiment of the filters shown in FIG. 1C, filters Delta (Δ) and Sigma (Σ) continuing in dashed line, and as modified in their synthesis, continuing in solid line, modifications alternatively accounting for the modifications represented by the filter factors G

and F, as shown in FIG. 2C, used in explaining the invention.

FIG. 3B is a set of magnitude-(db)-versus-frequency-(log scale) response curves of the transfer characteristics of a specific embodiment of the filter shown in FIG. 1C, having characteristics similar to those in FIG. 3A, showing first alternative modifications, used in explaining the invention.

FIG. 3C is, a set of magnitude-(dC)-versus frequency-(log scale) response curves of the transfer characteristics of the specific embodiment of the filters shown in FIG. 1A, having characteristics similar to those shown in FIG. 2C, showing the modifications therein that are the consequences of the alternative modifications shown in FIG. 3B, used in explaining the invention.

FIG. 4A is a set of magnitude-(dB)-versus-frequency-(log scale) response curves of the transfer characteristics of a specific embodiment of the filters shown in FIG. 1C, having characteristics similar to those shown in FIG. 3A, showing second alternative modifications, used in explaining the invention.

FIG. 4B is a set of magnitude-(dB)-versus-frequency-(log scale) response curves of the transfer characteristics of a specific embodiment of the filters shown in FIG. 1A, having characteristics similar to those shown in FIG. 2C, showing the modifications therein that are the consequences of the alternative modifications shown in FIG. 4A, used in explaining the invention.

FIG. 4C is a set of magnitude-(BB)-versus-frequency-(log scale) response curves of the transfer characteristics of a specific embodiment of the filters shown in FIG. 1C, having characteristics similar to those shown in FIG. 3A, showing third alternative modifications, used in explaining the invention.

FIG. 5A is a set of magnitude-(dB)-versus-frequency-(log scale) computer-generated response curves of the transfer characteristics of the Delta filter shown in FIG. 1C, having characteristics similar to those shown for the Delta filter in FIG. 3A, showing in dashed line the diffraction-computation specification, and in solid line the approximation thereto, with modification, computed for the synthesis via a specific sequence of biquadratic filter elements, used in explaining the invention.

FIG. 5B is a set of delay-versus-frequency-(log scale) computer-generated response curves of the transfer characteristics consequent to the magnitude characteristics of FIG. 5A, with a biquadratic-synthesis curve (minimum phase) shown in solid line, used in explaining the

FIG. 5C is a set of magnitude-(dB)-versus-frequency-(log scale) computer-generated response curves of the transfer characteristics of the Sigma filter shown in FIG. 1C, characteristics similar to those shown for the Sigma filter in FIG. 3A, showing in dashed line the diffraction-computation specifications, and in solid line the approximation thereto, with modifications, computed for the synthesis via a specific sequence of biquadratic filter elements, used in explaining the invention.

FIG. 5D is a set of delay-(vs)-versus-frequency-(log scale) computer-generated response curves of the transfer characteristics consequent to the magnitude characteristics of FIG. 5A, with a biquadratic-synthesis curve shown in solid line, used in explaining the invention.

FIG. 6 is a block diagram of a specific embodiment of a circuit illustrating sequences of biquadratic filter elements to obtain the solid line curves of FIG. 6A through FIG. 6D.

FIG. 7 is a schematic diagram illustrating a specific embodiment of a biquadratic filter element.

FIG. 8A is a generalized block diagram illustrating a specific embodiment of a shuffler-circuit inverse formatter to produce binaural earphone signals from signals intended for loudspeaker presentation.

FIG. 8B is a generalized block diagram of the same embodiment illustrated in FIG. 8A, wherein the difference-sum forming networks are each represented as single blocks.

FIG. 9 is a generalized block diagram illustrating a specific embodiment of a multiple shuffle-circuit formatter functioning as a synthetic head.

FIG. 10A is a generalized block diagram illustrating a specific embodiment of a reformatter to convert signals intended for presentation at one speaker angle (e.g., ±30°) to signals suitable for presentation at another speaker angle (e.g., ±15°), employing two complete shuffle-circuit formatters.

FIG. 10B is a generalized block diagram illustrating a specific embodiment of a reformatter for the same purpose as in FIG. 10A, but using only ne shuffle-circuit formatter.

FIG. 11 is a generalized block diagram illustrating a specific embodiment of a reformatter to convert signals intended for presentation via one loudspeaker layout to signals suitable for presentation via another layout, particularly ,one with an off-side listener closely placed with respect to one of the loudspeakers.

FIG. 12A is a set of transfer function curves plotted for an incidence angle of 30° and for a particular artificial head.

FIG. 12B is a set of transfer function curves plotted for an incidence angle of 30° and for a particular artificial head and for a 0° angle of incidence.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

FIG. 1A is a generalized block diagram illustrating a specific embodiment of a stereo audio processing system 50. The stereo system 50 comprises an artificial head 52 which produces two channels of audio signals which are coupled to a lattice network 54, as shown. The signals from the artificial head 52 may be coupled to the network 54 by first recording the signals and then reproducing them and coupling them to the network 54 at a later time. The artificial head 52 comprises a physical dummy head, which may be a spherical head in the illustrated embodiment, including appropriate microphones 64, 66. The artificial head may also be a replica of a typical human head using head dimensions representative of middle values for a large population. The output of the microphones 64, 66 provide audio signals having head-related transfer functions imposed thereon. The lattice network 54 provides crosstalk and naturalization compensation thereby processing the signals from the artificial head 52 to compensate for actual acoustical propagation path and head-related distortion.

The artificial head may alternately comprise a natural, living head whose ears have been fitted with miniature microphones, or it may alternately comprise a synthetic head. The synthetic head, to be described in detail at a later point in connection with FIG. 9, comprises an array of circuits simulating the signal modifying effects of head-related diffraction for a discrete set of source signals each designated a specific source bearing angle. The signals from such a head, or alternate, are each coupled to the network 54 which comprises filter cir-

cuits (S'G) 72, 74, crosstalk filters (A'F) 76, 78, and summing circuits 80, 82, configured as shown. The outputs of the network 54 are coupled to the loudspeakers 60 and 62, which are placed at a bearing angle (typically ±30°) for presentation to a listener 84, as shown. In one embodiment of the system 50, the summed signals at the summing circuits 80 and 82 may be recorded and then played back in a conventional manner to reproduce the processed audio signals through the loudspeakers 60 and 62.

An alternative embodiment of a stereo audio processing system is illustrated in generalized block diagram form in FIG. 1B. In the embodiment of FIG. 1B, the stereo audio processing system 100 comprises an artificial head 102 or alternative heads as indicated above in connection with FIG. 1A. The artificial head 102 is coupled, either directly or via a record/playback system to a compensation network 140 which comprises a crosstalk cancellation network 120 and a naturalizing network 130. The crosstalk cancellation network 120 comprises two crosstalk circuits 122 and 124 which impose a transfer function $C=-A/S$, where S is the transfer function for the acoustical propagation path characteristics from one loudspeaker to the ear on the some side, and A is the transfer function for the propagation path characteristics to the ear on the opposite side, as shown.

Each crosstalk circuit 122, 124 is substantially limited to frequencies substantially below ten kilohertz by low pass filters 121 and 123 with response characteristic F having cutoff frequency substantially below ten kilohertz. The output of the crosstalk filter circuits 121, 123 is summed with the output modified by the filters (G) 110, 112, by the summing circuits 126, 128, of the opposite channel, as shown. The resulting signals are coupled respectively to crosstalk correction circuits 132 and 134 which impose a transfer function of $1/(1-C^2)$. The resulting signals are coupled to the naturalization circuits 136 and 138 which impose a transfer function of 1/S, as shown. The output of the network 130 is then coupled, optionally via a recording/playback system, to a set of loudspeakers 140 and 142 for presentation to the ears 143, 145 of a listener 144, as shown.

FIG. 1C is a generalized block diagram of another alternative embodiment of a stereo audio processing system. The stereo audio processing system of FIG. 1C comprises an artificial head 151 comprising two microphones 152, 154 for generating two channels of audio signals having head-related transfer functions imposed thereon. A synthetic head, which is described in greater detail hereinafter with reference to FIG. 9, may alternatively be used. The audio signals from the artificial or synthetic head 151 are coupled, either directly or via a record/playback system, to a shuffler circuit 150, which provides crosstalk cancellation and naturalization of the audio signals.

FIG. 1D is a generalized block diagram of an alternative embodiment of a stereo audio processing system in accordance with the invention. The stereo audio processing system of FIG. 1C comprises an artificial head (including a real synthetic head) 151*a* comprising microphones 152*a*, 154*a* for generating two channels of audio signals having head related transfer functions imposed thereon. An equalization network 157, and another 159 are coupled to the audio outputs of the microphones 152*a*, 154*a* to provide equalization for the inputs to a cross-talk compensation network 150*a*. The equalization networks 157, 159 may also be coupled to the outputs of the crosstalk compensation network 150*a* to provide equalization of summed signals from a set of summing circuits 166*a*, 170*a* to be then coupled to the loudspeaker 172, 174.

The shuffler circuit 150*a* comprises a direct crosstalk channel 155*a* and an inverted crosstalk channel 156*a* which are coupled to a left summing circuit 158*a* and a right summing circuit 160*a*, as shown. The left summing circuit 158*a* sums together the direct left-channel audio signal and the inverted crosstalk signal coupled thereto, and couples the resulting sum to a Delta (Δ) filter 162*a*. The right summing circuit 160*a* sums the direct right-channel signal and the direct crosstalk left channel signal and couples the resulting sum to a Sigma (Σ) filter 164*a*. The output of the Delta filter 162*a* is coupled directly to a left summing 166*a* and an inverted output is coupled to a right summing circuit 170*a*, as shown. The output of the Sigma filter 164 is couple directly to each of the summing circuits 166*a* and 170*a*, as shown. The output of the summing circuits 166*a* and 170*a* is coupled, optionally via a record/playback system to a set of loudspeakers 172 and 174 arranged with a preselected bearing angle ϕ for presentation to the listener 176. Equalization circuit 157, 159 may be utilized alternatively between the summing circuits 166*a*, 170*a* and the loudspeakers 172, 174. The specific nature of the equalization and crosstalk compensation networks is discussed in detail hereinafter.

Each of the three alternative embodiments of FIG. 1A, 1B and 1C may be show to be equivalent. For the purposes of explaining the overall functioning of these configurations, let the filters F and G of FIGS. 1A and 1B be regarded as nonfunctioning, i.e., to have a frequency-independent transmission function of unity. (The purpose and design of these filters or alternative equivalents will be described in detail hereinafter). Then, if the transfer function through the direct path (through G) in FIG. 1B is computed, it is found to be $(1/S)/(1-C^2)$, equivalent to $S'=S/(S^2-A^2)$, to obtain a loudspeaker signal. Similarly, if the transfer function through the cross path (through F) is computed, it is found to be $(C/S)(1-C^2)$, equivalent to $A'=A/(S^2-A^2)$, to obtain a loudspeaker signal. These S' and A' transfer functions are the same functions used in FIG. 1A, and the same result would have been obtained if the F and G symbols had been carried along in the computation. The equivalence may be extended to FIG. 1C by requiring the Delta filter to be equal to $(S'-A')/2$ and requiring the Sigma filter to be equal to $(S'+A')/2$, which are $1/(S-A)$ and $1/(S+A)$, respectively, and there is little difficulty in carrying the F an G symbols through the derivation also.

Thus, an explanation of the functioning of any one of these embodiments will illustrate the functioning of them all. Referring to FIG. 1B, for example, where the acoustic-path transfer functions A and S are explicitly shown, it may be seen that the left ear signal at L_e 143 is derived from the signal at the microphone 114 via the transfer function $S^2/(S^2-A^2)$ involving path S, to which must be added the transfer function $-A^2/(S^2-A^2)$ involving path A, with the result that the transfer function has equal numerator and denominator and is thus unity. However, a corresponding analysis shows that the transfer function from the signal at the microphone 116 to the same ear, L_e 143 is $AS/(S^2-A^2)$ to which must be added $-AS/(S^231\ A^2)$, thus obtaining a null transfer function. This analysis illustrates crosstalk cancellation whereby each ear receives only the signal

intended for it despite its being able to hear both loudspeakers.

The embodiment of FIG. 1B, except for the F and G filters, was described by M. R. Schroeder in the American Journal of Physics, vol. 41, pp. 461–471 (April 1973), "Computer Models for Concert Hall Acoustics," FIG. 4, and later in the Proceedings of the IEEE, vol. 63, p. 1332–1350 (September, 1975) "Models of Hearing," FIG. 4. Earlier equivalent versions may also be seen in B. S. Atal and M. R. Schroeder, "Apparent Sound Source Translator," U.S. Pat. No. 3,236,949 (Feb. 26, 1966).

However, the embodiment of FIG. 1B will be inoperative if the various filter functions specified therein cannot be realized as actual signal processors. The question of realizability may be examined with the help of FIG. 2A and FIG. 2B, plots of the acoustic transfer functions S and A in magnitude and phase, respectively, for a spherical-model head. Plots for a more realistic model will differ from these only in details not relevant to realizability. Schroeder taught that the filter $C=-A/S$ would be realizable, having a magnitude sloping steeply downward with increasing frequency, and similarly for the phase, indicating a substantial delay. The corresponding finite impulse response calculated by Fourier methods would show a characteristic pulse shape substantially delayed from the time of application of the impulse. The fulfillment of this causality condition is of the essence of realizability. Such an impulse response may be realized as a transversal filter. Schroeder saw that the filter C^2 would also be realizable as a transversal filter, and that placement of C^2 in a feedback loop would produce the realization of $1/(1-C^2)$. The remaining filter, 1/S, however, would not be directly realizable because Schroeder's data to FIG. 2B, showed 1/S to exhibit a rising phase response being indicative of an advance, with calculation by Fourier methods showing a characteristic pulse response beginning prior to the application of the impulse. Nevertheless, it was realized that providing a frequency-independent delay that would be equal in the two loudspeaker channels would be harmless, so that a transversal-filter realization employing augmented delay would be satisfactory for 1/S.

The filter S' and A' of FIG. 1A have the transfer functions shown plotted in FIG. 2C for magnitude and in FIG. 2D for phase, from spherical-model calculations. Specific curves for S' and A' are represented by the solid-line curves with dashed-line continuation, while the solid line continuations show modifications imposed by the filter factor G, forming S'G, and imposed by the filter factor F forming A'F, the filters shown in FIG. 1A. However, the corresponding phase modifications are not shown in FIG. 2D, such further information not being required at this point.

It may be seen from these unmodified curves that the S' and A' filters are realizable because of the steep downward slopes with increasing frequency in the phase, indicating abundant delay to allow realization by transversal filters. Of course, if more delay were needed for that purpose, it would be harmless to provide equal increments in delay for each. In the configuration used by Schroeder and Atal, the filters to be realized are more nearly directly related to measurable data, S and A, and one may always proceed with the greater confidence the closer one stays to measured data in its original form. Nevertheless, the requisite filters are realizable, so that FIGS. 1A and 1B show equally acceptable configurations.

The rather large amounts of delay involved in the filters for both of the configurations of FIG. 1A and FIG. 1B, however, make them awkward for realization by means other than transversal filters or other devices capable of generating longer delays. Other means of realization, or synthesis, are much less troublesome and expensive if the filters to be synthesized are of the kind known as "minimum phase" because then simpler network structures may be used with efficient, more widely-known synthesis techniques. Minimum-phase filters have the property that the phase response may be calculated directly from the logarithm of the magnitude of the transfer function by a method known as the Hilbert transform. If the transfer function is not of minimum phase, the calculation results in only a part of the phase response, leaving an excess part that is the phase response of an all-pass factor in the transfer function. Although many examples of all-pass filters are known, the synthesis of the phase response of an arbitrarily-specified all-pass filter is not as well developed an art as the synthesis of minimum-phase filters.

It is known in the art that the excess phase in the transfer functions A and S is nothing more than a frequency-independent delay (or advance). Thus, the Schroeder filters C and 1/S could have been realized as minimum-phase filters together with a certain frequency-independent increment in delay, since products and ratios of minimum-phase transfer functions are also of minimum phase. However, it does not follow that $1-C^2$ would be of minimum phase. Thus, the phase status of A' and S' does not follow. The difference between two properly-chosen, minimum-phase transfer functions is one means of synthesizing an all-pass transfer function.

However, it is one aspect of the invention to teach the use of minimum-phase filter synthesis in these systems. The inventors have been able to show that the transfer functions $S+A$ and $S-A$ have excess phase that is nothing more than a frequency-independent delay (or advance). Since the product of these is S^2-A^2, all of the filters considered thus far may be synthesized as minimum-phase filters, together with appropriate increments in frequency-independent delay. This provides a distinct advantage since such augmentation is available through well-known means.

It is a further aspect of the invention to teach limiting the frequency response of the crosstalk cancelling filters A' to form A'F. The modification shown as the solid-line continuation in FIG. 2C illustrates the general form of such modifications delegated to the filter function F. The reason for limiting frequency response is that cancellation actually takes place at the listener's ears and it is reasonably exact in a region of space near each ear, a region that is smaller for the shorter wavelengths. Thus, if the listener should turn his head, his ear will be less seriously transported out of the region of nearly exact cancellation if the cancellation is limited to the longer wavelengths. Schroeder reports some 10° as the maximum allowable rotation, and some 6 inches as the maximum allowable sideways movement for his system. It is a teaching of this invention that limiting the response of the crosstalk cancelling filter to a frequency substantially below 10 KHz will still allow accurate image portrayal over a wide enough frequency band to be quite gratifying while allowing the listener to move over comfortable ranges without risking serious impair-

ment of the illusion. Experiments with an embodiment of the system illustrated in FIG. 1C confirm the correctness of this teaching.

The solid-line extension for curve S' in FIG. 2C illustrates one possible effect to be produced by the filter G of FIG. 1A and FIG. 1B. When the acoustic transfer functions are determined from the spherical model of the head, as used here for illustration, then the undulations determined for S' will not be the same as they would be for a more realistic model, especially at the higher frequencies. In accordance with the invention, the filter will not simulate the details of these undulations above a certain frequency. However, there is another reason not to simulate the higher-frequency undulations: listeners' heads will vary in ways that are particularly noticeable in measurements at the higher frequencies, specially in the response functions attributed to the pinna. Thus, above a certain frequency, it would not be possible to represent these undulations correctly, except for a custom-designed system for a single listener. A correct simulation of these undulations will, however, affect only the tone quality at these higher frequencies, frequencies for which the notion of "tone" becomes meaningless. It is sufficient to obtain the correct average high-frequency level, and dispense with detail. The solid-line extension of S' in FIG. 2C illustrates filter characteristics for one embodiment of the invention, and is characteristic of a system, as illustrated in FIG. 1C, which the inventors have constructed and with which they have made listening tests.

It is therefore to be seen that there are two reasons for limiting the crosstalk cancellation to frequency ranges substantially less than 10 kHz. The first reason is to allow a greater amount of listener head motion. The second reason is a recognition of the fact that different listeners have different head-shape and pinna (i.e., small-scale features), which manifest themselves as differences in the higher-frequency portions of their respective head-related transfer functions, and so it is desirable to realize an average response in this region.

Plots of the magnitude of the transfer functions Delta of FIG. 1C, namely $1-/(S-A)$, add of Sigma, namely $1/(S+A)$, are shown in solid line in FIG. 3A. There, the dashed-line continuation shows the transfer function specified in terms of S and A in full for the spherical model of a head, and the solid-line shows the transfer function approximated in the system of FIG. 1C. The consequence of the modification illustrated in FIG. 3A is, in fact, the modification illustrated in FIG. 2C. The means whereby these transfer functions were realized will be discussed at a later point. It is seen that the modification in FIG. 3A consists in requiring a premature return to the high-frequency asymptotic level (-6 dB), premature in the sense of being completed as soon as possible, considering economies in realization, above about 5 KHz.

The curve Delta in FIG. 3A shows an integration characteristic, a -20 dB-per-decade slope that would intercept the -6 dB asymptotic level at about 800 Hz, with a beginning transition t asymptotic level that is modified by the insertion of a small dip near 800 Hz, and a similar dip near 1.8 KHz, after which there relatively narrow peak characteristic at about 3.3 KHz rising some 7 dB above asymptotic, falling steeply back to asymptotic by about 4.5 KHz, followed by a small dip near 5 KHz, after which there is a rapid leveling out (solid-line continuation), at higher frequencies towards the asymptotic level. The curve Sigma in FIG. 3A shows a level characteristic at low frequencies that lies at the asymptotic level, followed by a gradual increase that reaches a substantial level (some 4 dB) above asymptotic by 800 Hz and continues to a peak at about 1.6 KHz at some 9.5 dB above asymptotic, after which there is a steep decline to asymptotic level at about 2.5 KHz, a small dip at about 3.5 KHz, followed by a narrow peak of some 6 dB at about 5.0 KHz, followed by a relatively steep decline to reach asymptotic level at about 6.3 KHz that is modified (solid-line continuation), beginning at about 6.0 KHz, to begin a rapid leveling out to the asymptotic level at higher frequencies.

The system of FIG. 1C also included a high-pass modification of these curves at extreme low frequencies, primarily to define a low-frequency limit for the integration characteristics of the Delta curve. The same high-pass characteristic is used for Sigma also, for the sake of equal phase fidelity between the two curves. Although a 35-Hz high-pass corner was chosen, in common, any in the range of approximately 10 Hz to 50 Hz would be very nearly equally satisfactory.

It is a teaching of this invention that these curves may be modified to approximate Delta and Sigma in a variety of ways, described below as alternative treatments of specifications of F and G for specific purposes. It is to be understood, however, that other modifications that result in curves following generalized approximations to the curves of FIG. 3A, or any of the curves thereafter, including approximations to the high-frequency trends, whether for the spherical-model head, or replica of a typical human head, or any other model, and including consequences of such generalized approximations for the filters of FIG. 1A and FIG. 1B, fall within the teachings of this invention.

The curves shown in FIG. 3B illustrate means of obtaining an alternate G-filter effect mentioned above. It is seen that the solid-line extension for Delta is made to join with the solid-line curve for Sigma as soon as reasonable after 5 KHz, but that the Sigma curve is unmodified. Thus the difference between the two curves quickly approaches null, as shown in FIG. 3C by the trend in A'F towards minus infinity decibels. Thus F is as before, but it is also seen that S'G is the same as S', i.e., G is unity. As mentioned before, this alternative would be useful in custom-designed formatters.

Another alternative treatment of G is illustrated in FIG. 4A. There, the premature return to a high-frequency level is to a level some 2 dB higher than asymptotic. The result is an elevated high-frequency level for S'G, as illustrated in FIG. 4B, while A'F shows the same high-frequency termination as previously indicated.

Inspection of FIG. 4A suggests a lower-frequency opportunity for premature termination to a high-frequency level, namely at about 2.5 KHz. By forcing the Delta and Sigma curves to follow the same function above such frequency, the cut-off frequency for low-pass filter F will in effect, be determined to lie at about 2.5 KHz, while the character of G will be determined by the alternative chosen for the character of the common function to be followed above 2.5 KHz. Restriction of the crosstalk cancellation to such low frequencies will make the imaging properties more robust (i.e., being less vulnerable to listener movement). The price to be paid for such augmented robustness is, of course, a diminishment in imaging authenticity.

However, a more general means to limit the frequency range of crosstalk cancelling, one more general

than the ad hoc process of looking for a propitious opportunity indicated by the curve shapes is illustrated in FIG. 4C. Indicated in FIG. 4C as a solid line is an approximation departing from the full specification, departures covering a broad range of frequencies, beginning with small departures at the lower frequencies, undertaking progressively larger departures at higher frequencies. Useful formatters may be constructed by such means, useful particularly to provide a more pleasing experience for badly-placed listeners that might thus perceive an untoward emphasis upon certain frequencies.

The specific filter responses used in constructing a test system as shown in FIG. 1C are illustrated in FIGS. 5A through 5D. These FIGS. 5A–5D show computer-generated plots of the spherical-model diffraction specifications in dashed line and plots of the accepted approximations in solid line. A computer was programmed to make the diffraction calculations and form the dashed line plot. However, it was also programmed to calculate the frequency response of the combination of filter elements to be constructed in realizing the filters and in making the solid-line plots. Then, the operator adjusted the circuit parameters of the filter elements to obtain close agreement with the diffraction calculations up to about 5 KHz. The filter thus designed was chosen to be a minimum-phase type. It was found that it is possible to obtain a simultaneous match for both the amplitude and the phase response except for an excess phase corresponding to nothing more than a

frequency-independent delay (or advance). Since filters $1/(S-A)$ and $1/(S+A)$ were being approximated, these were thus established as of minimum phase, at least over the frequency range explored.

FIG. 5A illustrates the extent of agreement between diffraction specification and accepted design for the magnitude of Delta, plotted in decibels versus frequency (log scale), and FIG. 5B illustrates the simultaneous agreement in phase. The latter is actually a plot of phase slope, or frequency-dependent delay in microseconds, versus the same frequency scale. Agreement in phase slope is at least equal in significance as agreement in phase, but is of advantage in sensing a disagreement in frequency-independent delay (or advance), and such uniform-with-frequency discrepancies were indeed found. Such discrepancies were found to be the same for both the Delta and Sigma filters and could thus be suppressed in the filter design. FIGS. 5C and 5D illustrate, respectively, curves similarly obtained for the Sigma filter.

Recordings have been made with an artificial head, and the recordings processed with a novel crosstalk canceller according to the invention embodying the filter-response curves of FIGS. 5C and 5D. The artificial head was a commercially available Neumann KU-80, whose microphones provide accurate ear-canal-entrance signals. Generally, with in this system the processed recordings are quite good, however, there can be a few instances in which the processed recordings sound somewhat like an ordinary stereo recording, lacking the full spatial envelopment except perhaps at low frequencies. In addition, in these instances the images that seemed largely confined to the space between loudspeakers, and, in the worst of these instances, seeming to avoid placing images near the center of that space. Listening to the unprocessed recordings, on the other hand, also showed the faults of these few instances consistently with the images tending to cluster near the loudspeakers even more severely than in ordinary stereo recording.

Investigation revealed that these few instances of results that were less than satisfactory could be traced to a common acoustic characteristic in the listening environment. In seeking to simulate a consumer-type environment, rooms had been mostly chosen that were somewhat reverberant. However, in some of these listening setups, the loudspeakers had been placed so that reflecting acoustic paths were allowed that differed from the direct acoustic paths, loudspeakers to ears, by delay amounts of up to a millisecond or so. Such competing paths, when of significant intensity and falling within the same delay range as occupied by the crosstalk-canceling signal, can spoil part of the cancelling effect. The rooms in which good results had been obtained were also reverberant, but the good result could be traced to a more fortunate loudspeaker placement, one sufficiently distant from reflecting surfaces to avoid these approximately one to two millisecond delay reflection paths.

Recordings that had been made with the *Aachener Koof* (AK), an artificial head made by Head Acoustics, GmbH, of Aachen, Germany were also processed with the novel crosstalk processing of the invention. These recordings had been previously equalized with circuits supplied by the maker to correct the microphone signals to provide a flat frequency response with reference to a plane wave incident upon the front of the head, an incidence angle of 0°. Upon listening to the unprocessed recordings, they showed an excellent normal stereo effect characterized by the common stereo condition of a smooth spread of images in the space between loudspeakers, including a natural tendency to place images somewhat outside this space, an overall stereo quality not typically attained by ordinary stereo recordings. Moreover, when the recordings from this head were crosstalk processed, hey fully satisfied every expectation as to full spatial envelopment, precise imaging to the front, to the extreme sides, behind, and in elevation.

Under unfavorable conditions (early reflecting paths), the processed AK recordings showed a degradation that was only moderate, retaining a stereo quality that was always excellent, always noticeably better than any ordinary stereo recording. This improved characteristic of relative insensitivity to listener-space acoustics is one of substantial utility. An analysis presented hereinafter leads to an optimal equalization practice to ensure this characteristic.

The principle technical effect of requiring the equalization for the artificial head to be a part of the head, not be a part of the crosstalk-cancelling filter, is to simplify the crosstalk-canceling filters by removing a common equalization factor and placing it on the head side of the head crosstalk-canceller interface. This provides an opportunity to make the design of the crosstalk cancelling filter be independent of the artificial head and to orient its design to suit the listener's head. This would be appropriate because it is the listener's head that participates in the acoustic crosstalk process that is to be cancelled. This alternate approach clarifies the role of the equalization to remove those frequency characteristics of the artificial head that would be essentially repeated, but should not be, in the listener's head. These are the resonances of the cavities in the external ear, the pinna, and, if included in the artificial head, the ear canal.

One aspect of the invention comprises optimizing equalization to provide a specific combination of free-field signals to be used for specific incidence angles, and to specify these angles in relation to the angles to be used for loudspeaker placement, which combination is to be equalized to make for a flat microphone-signal response specifically for that combination.

A detailed discussion of the basis for this equalization begins with reference to the previously defined Σ function defined to be equivalent to 1/(S+A) and the Δ function defined to be equivalent to 1/S−A. The terms Σ' is defined as equivalent to S+A, and Δ's as S−A. Since these and their reciprocals are of minimum phase, their phases constitute a redundant specification calculable by Hilbert transform and need not be specified, and their transfer functions are to be simulated by minimum-phase filters. Thus we deal with $|\Delta'|$ and $|\Sigma'|$ for analysis. . These can be expressed in terms of $|S|$, $|A|$, and cos $\omega\tau$, the last being the cosine of interaural phase (written as the product of angular frequency with interaural phase delay), as follows:

$$|\Delta'|=(|A|^2+|S|^2-2|AS|\cos\omega\tau)^{\frac{1}{2}} \quad (6)$$

and

$$|\Sigma'|=(|A|^2+|S|^2+2|AS|\cos\omega\tau)^{\frac{1}{2}}$$

Thus, as has been seen, frequency-response plots of these functions would show a pattern of interleaved alternations in curves that swing between an upper envelope of

$$|\Delta',\Sigma'|_{max}=|S|+|A| \quad (5a)$$

$$|\Delta',\Sigma'|_{min}=|S|-|A| \quad (5b)$$

These alternating curves intersect one another along a locus for which the cosine is null, and this locus is

$$|\Delta',\Sigma'|_{rms}=(|A|^2+|S|^2)^{\frac{1}{2}}=[(|\Delta'|^2+|\Sigma'|^2)/2]^{\frac{1}{2}}. \quad (6)$$

Of course, where Δ' and Σ' are equal, there is no crosstalk, so that $|\Delta',\Sigma'|_{rms}$ may be referred to as a "null-crosstalk, crosstalk locus." Actually, zero crosstalk requires Δ' and Σ' to be equal in phase as well as magnitude, and this is approximated only after $|\Delta'|$ and $|\Sigma'|$ have tracked each other over an extended frequency interval. As expressed by the last equation, however, the curve defines an equalization reference, because its square is the total power-spectrum transmission to the two ears. Thus a function $E(\omega,\theta)$ may be

$$|E(\omega.\theta)|=|\Delta',\Sigma'|_{rms},$$

a function dependent upon frequency and incidence angle. Taking E to be of minimum phase, it ca be used to define a free-field equalization for a particular reference (incidence) direction, θ_0.

The equalized transfer function for the difference signal is designated °N:

$$°N=\Delta'/E(\theta_o), \quad (8a)$$

and designated, °P for the sum,

$$°P=\Sigma'/E(\theta_o). \quad (8b)$$

The reference direction has been taken to be 0° for the AK (Aachen head), but, for loudspeakers to be placed at ±30°, a 30° is more appropriate.

Transfer-function data for an incidence angle of 30° and for a particular artificial head are shown plotted according to the above equations in FIGS. 12A and 12B. The solid-line curve **520** labelled "difference" is a plot of 1/°N, while the solid-line curve **522** labelled "sum" is a plot of 1/°P, in FIG. 12A, and the upper dashed-line curve **524** is a plot of $1/|\Sigma|_{min}$, while the lower dashed line curve **526** is a plot of $1/|\Delta'\Sigma'|_{max}$ each similarly equalized. The solid-line curve **520** in FIG. 12*b* is a plot of $1/|E|$, while the dashed-line curve **532** is the equalization curve that would be used for a 0° angle of incidence. For the sake of clarity, the 3-dB displacement between these two curves has been retained. These data are for an artificial head constructed at CBS Laboratories under a contract to NASA.

Comparison between FIG. 3A and FIG. 12A shows a generally similar structure. The null-crosstalk contour that may be constructed in FIG. 3A upon the intersection points is, however, not flat because those curves have not been normalized against the equalization curve for that spherical-model, pinna-free, head. It is, nevertheless, essentially flat, compared to the contours plotted in FIG. 12B, so that with the crosstalk canceller based on the curves of FIG. 3A performs essentially as expected for a flat null-crosstalk contour. Thus, this canceller is suitable for use with an artificial head provided with free-field equalization.

The difference between the curves of FIG. 12B, with due regard for the 3-dB inserted difference, are seen to be small compared to the range of variation shown in FIG. 12B, totalling some 24 dB. Thus, a canceller based upon FIG. 3A only approximating one that might be modeled from data taken for our own heads, would not provide decisive evidence as to the aptness of either curve of FIG. 12B compared to the other. The large variations in FIG. 12B are typical of pinna resonances, since ear canal resonances had been largely excluded in the design of the head.

The curves **520, 522** of FIG. 12A differ from those of FIG. 3A in detailed ways that are typical of the ways in which actual heads differ, one to another, so that the curves of FIG. 3A, not showing so much idiosyncratic detail, stand a chance of suiting a wider variety of listeners' heads, better too than those of FIG. 12A. Thus, the teaching of the prior art, of modeling crosstalk-cancelling filters on a specific artificial head is not sound, in general, unless a "custom fit" to such a "listener's" head is desirable for some special application, e.g., documenting the differences between such a precise fit in comparison to a "looser fit" in the design of crosstalk-cancelling filters. For equalization, however, it is desireable for the equalization curve, as in FIG. 12B, solid line **530**, measured for a specific head, be used to equalize that same head. If this be done for each head to be considered for use as pickup heads, then the same crosstalk canceller from which such equalization had been excluded may be used with such heads interchangeably.

For the design of the crosstalk canceller to suit a wide variety of listeners' heads, it would be appropriate to obtain a fairly large collection of equalized data such as shown in FIG. 12A from a fairly large sample of heads, align their structures, i.e., the intersection points of their curves, points of maxima, etc., and determine a composite curve over sections between alignment points, a kind of structured average. Then, departures from the result-

ing curves, constructed on averaged positions for the alignment points, may be undertaken to provide a tolerance for motion on the part of a listener's head. The use of sum-and-difference data equalized as in FIG. 12A greatly facilitates such design efforts. It is contemplated that the invention covers use of such design procedures even if the canceller is to use lattice-arrayed filters, or other types of filters, since the lattice-array filters may be derived from, for example, shuffler-array filters.

In an illustrated embodiment of the instant equalization techniques and systems the free-field transmission functions A and S, for a specified angle of incidence, determined by measurement of an artificial head, are used to determine the magnitude of an equalization function as the square root of the sum of squares of the magnitudes of A and S. Furter, a pair of identical, minimum-phase filers **157, 159** (see FIG. 1A) simulating the reciprocal of this equalization function, are used to equalize the response from each of the ear microphones in that head, for such heads as are to be used in making binaural recordings that are to be reproduced through loudspeakers placed at said specified angles relative to the listener s head. An aspect of one of the illustrated embodiments further specifies that any crosstalk cancelling **150*a*** of said recording be designed to exclude such equalization and be designed to suit the loudspeaker locations relative to the listener's head, or a variety of such heads.

Another aspect of one of the illustrated embodiments includes the measurement of said transmission functions for artificial heads whose microphone signals had been already equalized to some other standard to determine equalization filters in the said manner either to replace the existing equalization or to supplement it.

FIG. 6 is a detailed block diagram illustrating a specific embodiment of the system of FIG. 1C. Operational amplifiers (op amps) of Texas Instruments type TI 074 (four amplifies per integrated-circuit-chip package) were used throughout. The insertion of input, high-pass filters (35 Hz corner) is not shown. In FIG. 6, input signals are coupled from inputs **154, 156** to summing circuits **158, 160** and each input is cross coupled to the opposite summing circuit with the right input **156** coupled through an inverter **162**, as shown. An integrator **172** is placed in a Delta chain **170** as required at low frequencies, While inverters **173, 182** are inserted in both Sigma and Delta chains **170, 180**. In these chains, a signal-inversion (polarity reversal) process happens at several places, as is common in op-amp circuits, and the inverters may be bypassed, as needed, to correct for a mismatch of numbers of inversions. The signals from the inverters **173, 182** are coupled to a series of BQ circuits (Bi-quadratic filter elements, also known as biquads) **174** and **184**. The resulting signals are thereafter coupled to output difference-and-sum forming circuits comprising summing circuits **190, 192** and aniinverter **194**.

As is generally known, biquads may be designed to produce a peak (alternative: dip) at a predetermined frequency, with a predetermined number of decibels for the peak (or dip), a predetermined percentage bandwidth for the breadth of the peak (or dip), and an asymptotic level of 0 dB at extreme frequencies, both high and low.

A specific embodiment of a suitable biquadratic filter element **200** is shown in FIG. 7. Other circuits for realizing substantially the same unction are known in the art. The circuit element **200** comprises an operational amplifier **202**, two capacitors **204, 206** and six resistors **208, 210, 212, 214, 216,** and **218** configured, as shown. With the circuit-element values shown, a peak at 1 KHz, of 10 dB height, and a 3 dB bandwidth of 450 Hz will be characteristic of the specific embodiment shown. Design procedures for such filter elements are well known in the art. Digital biquadratic filters are also well known in the digital signal-processing art.

The stereo audio processing system of the invention provides a highly realistic and robust stereophonic sound including authentic sound source imaging, while reducing the excessive sensitivity to listener position of the prior art systems. In the prior art systems, such as Schroeder and Atal, in which head-related transfer function compensation has been used, the entire audio spectrum (20 hertz to 20 kilohertz) was compensated and the compensation was made as completely accurate as possible. These systems produced good sound source imaging but the effect was not robust (i.e., if the listener moved or turned his head only slightly, the effect was lost). By limiting the compensation so that it is substantially reduced at frequencies above a selected frequency which is substantially below ten kilohertz, the sensitivity to the listener movement is reduced dramatically. For example, providing accurate compensation up to 6 kilohertz and then rolling off to effectively no compensation over the next few kilohertz can produce a highly authentic stereo reproduction, which is also maintained even if the listener turns or moves. Greater robustness can be achieved by rolling off at a lower frequency with some loss of authenticity, although the compensation must extend above approximately 600 hertz to obtain significant improvements over conventional stereo.

To obtain the binaural recordings to be processed, an accurate model of the human head fitted with carefully-made ear-canal microphones, in ears each with a realistic pinna may be used. Many of the realistic properties of the formatted stereo presentation are at least partially attributable to the use of an accurate artificial head including the perception of depth, images far to the side, even in back, the perception of image elevation and definition in imaging and the natural frequency equalization for each.

It may be also true that some subtler shortcomings in the stereo presentation may be attributable to the limitation in bandwidth for the crosstalk cancellation and to the deletion of detail in the high-frequency equalization. For example, imaging towards the sides and back seemed o depend upon cues that were more subtle in the presentation than in natural hearing, as was also the case with imaging in elevation, although a listener could hear these readily enough with practice. Many of the needed cues are known to be a consequence of directional waveform modifications above some 6 KHz, imposed by the pinna. It is significant that these cues survived the lack of any crosstalk cancellation or detailed equalization at such higher frequencies, a survival deriving from the depth of the shadowing by the head at such high frequencies so that such compensating means are less sorely needed.

The experience of dedicated "binauralists" is that almost any acoustical obstacle placed between 6-inch spaced microphones is of decided benefit. Such obstacles have ranged from flat baffles resembling table-tennis paddles, to cardboard boxes with microphones taped to the sides, to blocks of wood with microphones recessed in bored holes, to hat-merchant's manikins with microphones suspended near the ears. One may, of

course, think of spheres and ovoids fitted with microphones. Each of these has been found, or would be supposed with justice, to be workable, depending upon the aspirations of the user. The professional recordist will, however, be more able to justify the cost of a carefully-made and carefully-fitted replica head and external ears. However, any error in matching the head to a specific listener is not serious, since most listeners adapt almost instantaneously to listening through "someone else's ears." If errors are to be tolerated, it is less serious if the errors tend toward the slightly oversize head with the slightly oversize pinnas, since these provide the more pronounced localization cues.

This head-accuracy question needs to be carefully weighed in designing formatters that involve simulating the effect of a head directly, as for the synthetic head to be described hereinafter. One approach is to use measured head functions for these formatters. Fortunately, the excess delay in (S−A) and (S+A), the needed functions, is that of a uniform-with-frequency delay (or advance). The measurements, for most purposes, need be only of the ear signal difference and of the ear-signal sum, for carefully-made replicas of a typical human head in an anechoic chamber, and for most purposes only the magnitudes of the frequency responses need be determined. This is fortunate, since the measurement of phase is much more tedious and vulnerable to error. Such phase measurements as might be advantageous in some applications, need be only of the excess phase, i.e., that of frequency-independent delay, against an established free-field reference.

An example of direct head simulation would be that of a formatter to accept signals in loudspeaker format with which to fashion signals in binaural format (i.e., an inverse formatter). FIG. 8A illustrates a specific embodiment of a head-simulation inverse formatter **240** including a difference-and-sum forming network **242** comprising summing circuits **244**, **246** and an inverter **248** configured as shown. The difference and sum forming circuit **242** is coupled to Delta-prime filter **250** and a Sigma-prime filter **252**, the primes indicating that the filter transfer functions are to be S−A and S+A, instead of their reciprocals. The outputs of the Delta-prime and Sigma-prime filters is coupled, as shown, to a second difference and sum circuit **260**, as shown. The first appearance of an inverse formatter, or its equivalent may be found in Bauer, "Stereophonic Earphones and Binaural Loudspeakers," Jour. Acoust. Soc. Am., vol. 9. pp. 148–151 (April 1961), using separate S and A functions in approximation, showing a low-pass cutoff in A above about 3 KHz, and necessarily using explicit delay functions. See also Bauer, U.S. Pat. No. 3,088,997. It is an object of this aspect of the invention to improve upon Bauer by providing a more accurate head simulation, eliminating the low-pass cut for A, and avoiding the explicit use of delay by employing the shuffler configuration with Delta-prime and Sigma-prime filters. The use of faithful realizations of actual measured functions provides a further improvement. Since crosstalk cancellation is not a goal, there is no need for any kind of bandwidth limitation.

An accurate head simulator in this form is suitable for use with walk-type portable players using earphones. The conversion of binaurally-made, loudspeaker-format recordings back to binaural is highly suitable for such portable players. Questions of cost naturally arise in considering a consumer product, and particularly economical realizations of the filters are desirable and may be achieved by resorting to some compromise regarding accuracy and specifically using spherical model functions.

A block diagram of the inverse formatter **240** using an alternative symbol convention for the difference-and-sum-forming circuit is shown in FIG. 8B. Through the box symbol, the signal flow is exclusively from input to output. Arrows inside the box confirm this for those arrows for which there is no signal-polarity reversal but a reversed arrow, rather than indicating reversed signal-flow direction, indicates, by convention, reversed signal polarity. Also by convention, the cross signals are summed with the direct signals at the outputs.

The above conventions are used, for compactness, in making the generalized block diagram of a specific embodiment of a synthetic head **300** illustrated in FIG. 9. A plurality of audio inputs or sources **302** (e.g., from directional microphones, a synthesizer, digital signal generator, etc.) are provided at the top right each being designated (i.e., assigned) for specific bearing angle, here shown as varying by 5. increments from −90° to +90° although other arrays are possible. Symmetrically-designated input pairs are then led to difference-and-sum-forming circuits **304**, each having a Delta-prime output and a Sigma-prime output, as shown. Each Sigma-prime output is coupled to a respective Sigma-prime filter and each Delta-prime output is coupled to a Delta-prime filter, as shown. The Delta-prime outputs are summed, and the Sigma-prime outputs are summed, by summing circuits **306**, **308**, separately and the outputs are then passed to a difference-and-sum circuit **310** to provide ear-type signals (i.e , binaural signals). The treatment of the 0°-designated input is somewhat exceptional because it is not paired, and the Sigma-prime filter for it is $2S(0)=S(0°)+A(0°)$, determined for 0°, and its output is summed with that of the other Sigmas. In the diagram, ellipses are used for groups of signal-processing channels that could not be specifically shown.

In the synthetic head **300**, the Delta-prime and Sigma-prime filters may be determined by measurement for each of the bearing angles to be simulated, although for simple applications, the spherical-model functions will suffice. Economies are effected in the measurements by measuring only difference and sums of mannikin ear signals and in magnitude only, as explained above. A refinement is achieved by the measurement of excess delay (or advance) relative to, say, the 0° measurement. This latter data is used to insert delays, not shown in FIG. 9, to avoid distortions regarding perceptions in distance for the head simulation.

With regard to equalization, it is clear from the prior art that the purpose of earphone equalization is to restore the cavity resonance of the ear pinna disturbed by the placement of earphones on the ears so that the ear-canal sound is the same as if the soundwave had impinged in the uncovered pinna. Also of interest is making the pressure response of the ear drum be flat with respect to the electrical signals supplied to the earphone. Doctrines differ as to the soundfield that is to be simulated as impinging on the pinna, whether it is to be a diffuse field or to be a free, plane-wave soundfield.

That part of the prior art that specifies a free-field equalization also specifies 0° incidence. However, if crosstalk simulation is to be employed to simulate the sound from loudspeakers at ±30°, the earphone equalization should be designed for 30°. Similarly, if an artificial head, or electronic simulation thereof, is to be used to provide binaural signals equalized for a 30°

reference direction, then the earphone equalization should be designed for 30°.

Thus, earphone equalization, according to the invention, entails the use of probe microphones in the ear canals of a representative listener, or artificial head, for two cases, one wearing earphones whose signals are supplied from crosstalk-simulating circuits modeled on that same head that have a flat null-crosstalk locus, and the other with pinnas uncovered to a plane wave incident at the simulated angle, so that the earphone disturbance as the square root of the sum squares of A and S may be determined. The equalization filters are then constructed to correct this disturbance and used to filter the input signals into the earphones for reproduction.

The invention applies to the determination of equalization either as a replacement for a prior equalization that may be available or the earphones or as a supplement to such equalization. The invention also applies to equalization derived from structural averaging of data for a number of heads each measured in the manner stated hereinbefore.

Binaural synthesis may employ crosstalk simulating filters that have a flat null-crosstalk locus. It should be clear that, since lattice-array crosstalk simulating filters may be derived from shuffler-array sum-and-difference filters, a flat null-crosstalk-locus characteristic for the corresponding lattice filters is readily specified. This flat locus should be unmodified for the filters that simulate the same incidence angle that specifies the location of the loudspeakers. For the simulation of other incidence angles, the flat locus should be modified by the ratio of E functions, the ratio of that to be simulated to that for the loudspeaker locations, to serve as a specified equalization for simulating each of these other angles.

Since these crosstalk simulating filters will naturally be modeled after a specific representative head, the above equalization is equivalent to having provided the head with equalization as taught herein. The equalization functions specified in the previous paragraph may, of course, be merged with the characteristics of the simulating filters as may prove convenient, so as not to appear as distinct characteristics, without departing from the invention.

These equalization techniques and systems apply to the various audio applications recited in this application as well as to crosstalk cancellation and crosstalk simulation schemes, artificial-head microphone arrangements, and earphone equalization schemes found in the prior art.

Head simulation and head compensation used together provide another aspect of the invention, a loudspeaker reformatter. A specific embodiment of a loudspeaker formatter **400** in accordance with the invention is illustrated in FIG. 10A. The loudspeaker reformatter processes input signals in two steps. The first step is head simulation to convert signals intended for a specific loudspeaker bearing angle, say ±30°, to binaural signals, which is performed by an inverse formatter **403** such as that shown in FIG. 8B. The processing in the second step is to format such signals for presentation at some other loudspeaker bearing angle, say ±15° by means for a binaural processing circuit **404** such as that shown in FIG. 1C. The two steps may, of course, be combined, as is illustrated in FIG. 10B. An application of such a reformatter may exist in television stereo wherein it is very difficult to mount loudspeakers in the television cabinet so that they would be placed at bearing angles so large as ±30° for a viewer.

Another aspect of the invention provides loudspeaker reformatting for nonsymmetrical loudspeaker placements such as might be found in an automobile wherein the occupants usually sit far to one side. A nonsymmetrical loudspeaker reformatter **500** in accordance with the invention is illustrated in FIG. 11. Compensation for the fact that the listener **512** is in unusual proximity to one loudspeaker **516** is accomplished by the insertion of delay **502**, equalization **504** and level adjustment **506** for that loudspeaker. The delay and level adjustments are well known in tee prior art. However, a loudspeaker reformatter **508** provides equalization adjustment from head diffraction data for the bearing angle of the virtual loudspeaker **520**, shown in dashed symbol, relative to the uncompensated, other-side loudspeaker **514**. While a very good impression of the recording is ordinarily possible for such off-side listeners improved results can be obtained with such reformatting. Switching facilities may be provided to make the reformatting available either to the driver, or to the passenger, or to provide symmetrical formatting.

A specific embodiment of the stereo audio processing system according to the invention has been described for the purpose of illustrating the manner in which the invention may be made and used. It should be understood that implementation of other variations and modifications of the invention and its various aspects will be apparent to those skilled in the art, and that the invention is not limited by these specific embodiments described. It is therefore contemplated to cover by the present invention any and all modifications, variations, or equivalents that fall within the true spirit and scope of the basic underlying principles disclosed and claimed herein.

What is claimed is:

1. An equalization method for an audio processing system that generates compensated audio signals suitable for reproduction to a listener through a loudspeaker system, including source means for providing two channels of audio signals having head-related transfer functions imposed thereon, and compensation means for providing an inverse crosstalk in the audio signals to correct for the acoustic crosstalk characteristic of loudspeaker-to-listener ear transmission paths by employing a two port input, and two port output, cross-coupled filter system having transfer functions which approximately simulate acoustic transfer functions of the propagation paths from a loudspeaker to a first ear of the listener and from the loudspeaker to the second ear of the listener, said equalization method characterized by the steps of:

modifying signals at both ports of at least one of the input and the output of said compensation means by transmission of each signal through a filter that is essentially the same for each of the signals, said filter simulating an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of the squares of the magnitudes of said acoustic transfer functions.

2. The equalization method of claim 1, wherein said equalization transfer function is incorporated into the filters of the said compensation means to provide an equalization that is the equivalent of modifying at least one of the input and output signals of said compensation means.

3. The equalization method of claim 1, wherein said acoustic transfer functions are modified by division by said equalization transfer function, and said modified

functions are approximately simulated in said compensation means to provide an equalization that is the equivalent of modifying the signals of at least one of input and the output of said compensation means.

4. An audio processing system including equalization to simulate an acoustic process that imposes head-related transfer-function characteristics upon a plurality of audio signals, comprising:

source means for providing a plurality of audio signals, each designated as corresponding to a respective incidence direction of a plurality of incidence directions relative to a front-reference incidence direction;

a plurality of simulation means for imposing head-related transfer-function characteristics corresponding to each said designated incidence direction upon each respective signal from said source means, and each simulation means characterized by a two-port input, nd two-port output, cross-coupled filter means whose transfer function simulates the acoustic transfer functions for a source incidence direction to a listener's ear and for said source incidence direction to the listener's other ear, each simulating means including filters in feedback arrangements simulating approximations of said acoustic transfer functions, to produce a left-ear-designated signal and a right-ear-designated signal;

summing means for summing left-ear-designated signals together and for summing right-ear-designated signals together from the said plurality of simulation means to provide two combined outputs;

a plurality of equalization filters for simulating the reciprocal of an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of squares of the magnitudes of the said acoustic transfer functions determined for a reference incidence direction other than the front direction; and

means for modifying signals at least at one of the input and the output of each of said simulation means by transmission of each signal through one of the equalization filters that is substantially the same for each of the ports.

5. The system of claim 4, wherein the means for modifying also modifies the combined outputs of the summing means.

6. The system of claim 4, wherein said equalization transfer function is incorporated into the filters of each of said simulation means to provide an equalization that is the equivalent of modifying at least one of the input and the output of each of said simulation means.

7. The equalization method of claim 4, wherein the acoustic transfer functions are modified by division by said equalization transfer function, and are approximately simulated by said simulation means to provide an equalization that is the equivalent of modifying the signals of at least one of the input and the output of each respective simulation means.

8. An equalization system for a source of audio signals comprising two microphones mounted either in the two ears of an artificial head or in a close proximity to the ears of a person's head, each microphone generating an audio signal, said equalization system comprising:

a pair of equalization filters to modify in substantially the same manner each of the two signals obtained from the microphones, said equalization filters simulating the reciprocal of an equalization transfer function whose magnitude is proportional to the square root of the sum of squares of the magnitudes of acoustic transfer functions for a sound source incidence direction to a listener's ear and for said sound source incidence direction to the listener's other ear.

9. The equalization system of claim 8, wherein the microphones have been supplied with additional equalization other than that provided by said equalization filters such that the transfer functions to the listener's ears are not purely acoustic, and include the effects of said additional equalization, such that said equalization filters provide an equalization that is supplemental to said additional equalization.

10. An audio recording method for recording equalized audio signals suitable for reproduction to a listener through a loudspeaker system in conjunction with a crosstalk compensation circuit, the method comprising the step of:

providing two channels of audio signals having head related transfer function imposed thereon;

filtering the channels of audio signals by processing the respective signals through a filter means which simulates an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of the squares of the magnitudes of a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from a loudspeaker to a first ear of the listener and a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from the loudspeaker to a second ear of the listener to create equalized audio signals; and

recording the equalized audio signals suitably for subsequent reproduction.

11. An equalization system for earphones, comprising:

a designated pair of essentially identical earphones for which equalization is intended;

measurement means for determining the free-space acoustic transfer functions from a sound source to each of two predetermined corresponding points at two ears of a representative natural or artificial head;

signal source means for providing two essentially identical signals;

signal modifying means for imposing head-related transfer functions for a designated source direction upon said signals;

means for determining an acoustic transfer function from a free-space acoustic source to one ear from said source direction and an acoustic transfer function from the source to the other ear, and for determining a first equalization transfer function whose magnitude is the square root of the sum of squares of the magnitudes of the acoustic transfer functions;

means for determining an earphone equalization transfer function that is the first equalization transfer function divided by a second equalization transfer function,

filter means for simulating an approximation to said earphone equalization transfer function to modify two signals each in essentially the same manner,

means for coupling the two modified signals to the earphones.

12. An audio processing system that generates compensated and equalized audio signals suitable for reproduction to a listener through a loudspeaker system, comprising:

source means for providing two channels of audio signals having head related transfer functions imposed thereon;

compensation means having two input channels and two output channels for providing an inverse crosstalk in the audio signals to correct for the acoustic crosstalk characteristic of loudspeaker to listener ear transmission paths having a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from a loudspeaker to a first ear of the listener and a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from the loudspeaker to the second ear of the listener; and

filter means, coupled to the compensation means, for simulating an equalization transfer function whose magnitude is approximately proportional to the square foot of the sum of the squares of the magnitudes of said acoustic transfer functions.

13. A method of audio processing that generates compensated and equalized audio signals suitable for reproduction to a listener through a loudspeaker system, the method comprising the steps of:

providing two channels of audio signals having head related transfer functions imposed thereon;

providing an inverse crosstalk of the audio signals to correct for the acoustic crosstalk characteristic of loudspeaker to listener ear transmission paths, having a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from a loudspeaker to a first ear of the listener and a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from the loudspeaker to a second ear of the listener; and

simulating an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of the square of the magnitudes of said acoustic transfer functions.

14. An audio processing system that generates compensated and equalized audio signals suitable for reproduction to a listener through a loudspeaker system, comprising;

source means for providing two channels of audio signals having head related transfer functions imposed thereon;

compensation mans having two input channels and two output channels for providing an inverse crosstalk in the audio signals to correct for the acoustic crosstalk characteristics of loudspeaker to listener transmission paths having a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from a loudspeaker to a first ear of the listener and a transfer function which approximately simulates a free field acoustic transfer function of the propagation path from a loudspeaker to a second ear of the listener and wherein the free field acoustic transfer functions are modified by division by an equalization transfer function whose magnitude is approximately proportional to the square root of the sum of the squares of the magnitudes of said acoustic transfer functions to thereby provide equalization and crosstalk compensation.

* * * * *